Dialogic®

Dialogic® Vision™ 1000 Programmable Media Platform

User's Manual

January 2011 64-0400-04

www.dialogic.com

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Revision history

Revision	Release date	Notes
64-0400-04 Rev A	January 2011	BK, Dialogic® Vision™ 1000 Programmable Media Platform 5.1 and Dialogic® Vision™ 1000 Video Gateway 5.1
64-0400-03 Rev A	May 2010	BK, Dialogic® Vision™ 1000 Programmable Media Platform 5.0 and Dialogic® Vision™ 1000 Video Gateway 5.0
64-0400-02 Rev B	December 2009	BK, Dialogic® Vision™ VX Integrated Media Platform 4.2 and Dialogic® Vision™ CX Video Gateway 4.2
64-0400-02 Rev A	August 2009	BK, Dialogic® Vision™ VX Integrated Media Platform 4.2 and Dialogic® Vision™ CX Video Gateway 4.2
64-0400-01 Rev A	June 2009	BK, Dialogic® Vision™ VX Integrated Media Platform 4.1 and Dialogic® Vision™ CX Video Gateway 4.1
Last modified: 2011-01-13		

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1. Introduction

The *Dialogic® Vision™ 1000 Programmable Media Platform User's Manual* provides configuration, administration, and management information as well as application development guidelines for those who choose to use the Dialogic® Vision™ 1000 Programmable Media Platform. This manual assumes that you are familiar with the VoiceXML language and coding web applications.

For information about installing the Dialogic® Vision™ Server hardware, see the relevant hardware installation manual (listed in Related documentation).

Note: Product names have been changed since Dialogic® Vision™ 1000 Programmable Media Platform Release 5.0. The table below indicates terminology that was formerly associated with the products, as well as the new terminology by which the products are now known.

Former terminology	Current terminology
Dialogic® Vision™ VX Integrated Media Platform	Dialogic® Vision™ 1000 Programmable Media Platform Also referred to as "Programmable Media Platform"
Dialogic® Vision™ CX Video Gateway	Dialogic® Vision™ 1000 Video Gateway Also referred to as "Video Gateway"

The terms "Dialogic® Vision™ Server", "Vision Server", or "server" are used in this document to refer collectively or individually (depending on specific context) to the Dialogic® Vision™ 1000 Video Gateway or the Dialogic® Vision™ 1000 Programmable Media Platform.

2. Overview of the Dialogic® Vision™ 1000 Programmable Media Platform

Programmable Media Platform overview

The Programmable Media Platform is a VoiceXML media server that enables users to deploy video and speech-enabled applications in IP, ISDN, and ISUP networks. It provides the following functionality:

- Supports industry-standard VoiceXML 2.1 and the use of CCXML to control inbound and outbound dialing, call transfers, and conferencing.
- Supports the HTTP streaming of media files with caching on the server.
- Optionally provides in-band DTMF support.
- Optionally connects incoming faxes to a third-party T.38 server.
- Optionally provides fast call setup techniques to speed up 3G-324M call setup time.
- Produces detailed event logs with multiple information levels.
- Provides a web-based console, called the Dialogic® Vision™ Console (also referred to as "Vision Console" in this manual), to configure and manage the Programmable Media Platform.
- Provides an SNMP interface for monitoring application usage and server health.

The Programmable Media Platform is described in more detail in the following topics:

- Software components
- Signaling protocols and models
- Media capabilities
- Interfaces
- Fast call setup

Software components

The Programmable Media Platform contains the Call Server and the VoiceXML Subsystem.

The Call Server manages call control and transaction capabilities for all Programmable Media Platform models except IP-only audio.

The VoiceXML Subsystem includes the following software components:

Component	Description
VoiceXML Interpreter	Interprets VoiceXML dialogs. At startup, the VoiceXML Interpreter loads a media resource object named MIOSIP. This object provides the Media Resource Function with the ability to process media.
SSML Processor	Processes SSML requests for multimedia and text-to-speech. At startup, the SSML Processor loads an audio driver TCP object named ADTCP. This object serves as a link between the SSML Processor and the VoiceXML Interpreter.
Media Resource Function	Provides media processing including record, playback, and interfaces to speech recognition resources.

Signaling protocols and models

The Programmable Media Platform can implement the following signaling protocols:

- ISDN, which is available with the ISDN audio and ISDN video models.
- ISUP, which is available with the ISUP audio and ISUP video models. ISUP models can be configured for scalability and redundant capability.

The Programmable Media Platform uses SIP/RTP signaling to interface with the IP network. For more information, see ISDN models, ISUP models, and IP models.

Media capabilities

The Programmable Media Platform supports the following audio media formats:

Format	Encoding
Raw (headerless)	8 kHz 8-bit mu-law and A-law (G.711)
WAV (RIFF header)	8 kHz 8-bit mu-law and A-law, and 8 kHz 16-bit linear
Sun Audio (AU)	8 kHz 8-bit mu-law and A-law, and 8 kHz 16-bit linear
SPHERE (NIST header)	8 kHz 8-bit mu-law, and 8 kHz 16-bit linear

The ISDN video, ISUP video, and IP-only video models also support the video/3gpp MIME type.

The Programmable Media Platform supports the following codecs:

Codec type	Codec	Description
Audio	AMR	IETF RFC 3267; 3GPP TS 26.090, 26.101, and 26.073, version 5.3.0, 2004.
	G.711 A-law and mu-law	
	G.723.1	
	G.726	
	G.729 A	
	Comfort noise	IETF RFC 3389
Video	H.263	**Standard:** IETF RFC 2190, ITU-T Recommendation H.263, and 3GPP specifications TS.26.111, TS.26.911, TS.26.140. **Encoding format (Profile/Level):** Baseline level 10, 20, 30, 45. **Picture format:** QCIF Quarter Common Interchange Format (176 x 144) and CIF Common Interchange Format (352 x 288). **Frame rate:** 6 to 30 fps. Integer value only. 3G side is up to 15 fps. IP side is determined via SDP negotiation. **Encoding bit rate:** Up to 384 kbps. 3G side is 42 kbps. IP side is determined via SDP negotiation.
	H.263+	**Standard:** IETF RFC 2429, ITU-T Recommendation H.263, and 3GPP specifications TS.26.111, TS.26.911, TS.26.140. Picture format, frame rate, and encoding bit rate are the same as for H.263.

Codec type	Codec	Description
	H.264	**Standard:** IETF draft-ietf-avt-rtp-rfc3984bis-06.txt, ITU-T Recommendation H.264, and 3GPP specifications TS.26.111, TS.26.911, TS.26.140. **Encoding format (Profile/Level):** Baseline profile, level 1.0, 1b, 1.1, 1.2, 1.3, 2, 2.1, 2.2, 2.3, and 3. Packetization mode single-NAL and non-interleave. Levels 1.0, 1b, 1.1, and 1.2 support the same picture format, frame rate, and encoding bit rate as for H.263, except that CIF at 30 fps is not supported. Level 1.3 supports CIF and QCIF at 30 fps with a bit rate of 768 kbps. Levels 2, 2.1, 2.2, 2.3, and 3 support CIF at 30 fps with a bit rate of 2 Mbps.
	MPEG-4	**Standard:** IETF RFC 3016, ISO/IEC 14496-2:2004 and 3GPP specifications TS.26.111, TS.26.911, TS.26.140. **Encoding format (Profile/Level):** Simple profile level 0, 0b, 1, 2, and 3. MPEG-4 supports the same picture format, frame rate, and encoding bit rate as for H.263.
Fax relay	T.38	
Transparent media	NbUP	3GPP TS 29.415
	Clearmode	IETF RFC 4040

Interfaces

The Programmable Media Platform includes the following programmatic and management interfaces:

Interface	Description
VoiceXML	W3C standardized interface for creating user dialogs.

Interface	Description
Call Control Extensible Markup Language (CCXML)	W3C interface that enables the Programmable Media Platform to execute applications written in the W3C Working Draft of CCXML dated 29th June 2005 (http://www.w3.org/TR/2005/WD-ccxml-20050629). You can use CCXML to write applications that provide call control for the duration of a phone call, including call setup, monitoring, and teardown. You can also use CCXML applications to provide call routing and conferencing functionality. For information, see Managing CCXML applications, the *Dialogic® Vision™ CCXML Developer's Manual*, and http://www.w3.org/TR/2005/WD-ccxml-20050629.
Authorization and Usage Indication interface	XML-over-HTTP mechanism for access control over session establishment and gathering information for call detail reports. For information, see Overview of the Authorization and Usage Indication interface.
SIP interface	Signaling interface presenting calls or a method of establishing sessions for an application server. For information, see Overview of the SIP interface.
SNMP	Agent that provides enterprise MIBs to monitor server performance, view statistics, monitor a RAID server, and receive alarms. The master agent supports SNMPv1, SNMPv2c, and SNMPv3. For information, see the *Dialogic® Vision™ SNMP Reference Manual*.

Fast call setup

The Programmable Media Platform supports the following techniques for speeding up 3G-324M call setup time:

- Packed H.245 messages, which group independent H.245 messages together into a single NSRP command frame. This reduces the number of message round-trips, and thus reduces call setup time.
- Windowed Simple Retransmission protocol (WNSRP), an H.245 transport improvement technique that is standardized in ITU-T Recommendation H.324 and accepted into the 3G-324M standard by 3GPP.
- Media oriented negotiation acceleration (MONA), which unites the technologies for H.324 call setup acceleration under a common framework. MONA was approved by the ITU-T in August 2006, and is recommended in 3GPP Release 7 in TR 26.911.

Note: You must obtain the appropriate license to use one of these techniques. For information, see the *readme* file for this release.

The Programmable Media Platform does not support:

- The signaling preconfigured channel (SPC) MONA technique.
- Preconfigured channel media frames encapsulated in MONA signaling preference messages.

ISDN models

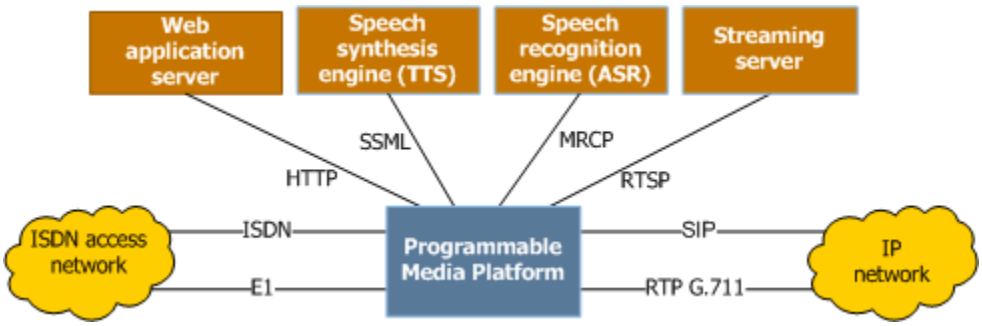
The Programmable Media Platform is available with an ISDN audio model and an ISDN video model.

ISDN audio model

In the ISDN audio model, the platform:

- Provides ISDN signaling.
- Provides SIP signaling.
- Supports the Call Control Extensible Markup Language (CCXML) for call control.
- Executes VoiceXML applications and their associated media processing functions.

The following illustration shows the ISDN audio model:



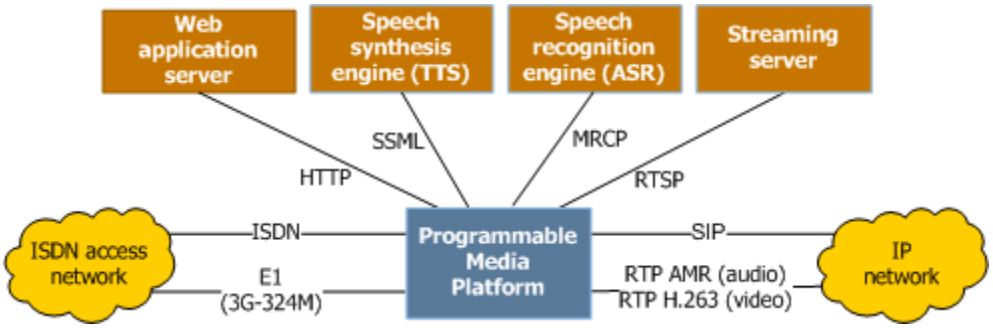
ISDN video model

In the ISDN video model, the platform:

- Provides ISDN signaling with 3G-324M.
- Provides SIP signaling.
- Provides the option to transcode between AMR and G.711.
- Supports the Call Control Extensible Markup Language (CCXML) for call control.
- Executes VoiceXML applications and their associated media processing functions.

The ISDN video model requires a mobile video device that supports 3G-324M.

The following illustration shows the ISDN video model:



ISUP models

The Programmable Media Platform is available with an ISUP audio model and an ISUP video model. Each of these models can be configured for scalability and redundant capability.

This topic describes the:

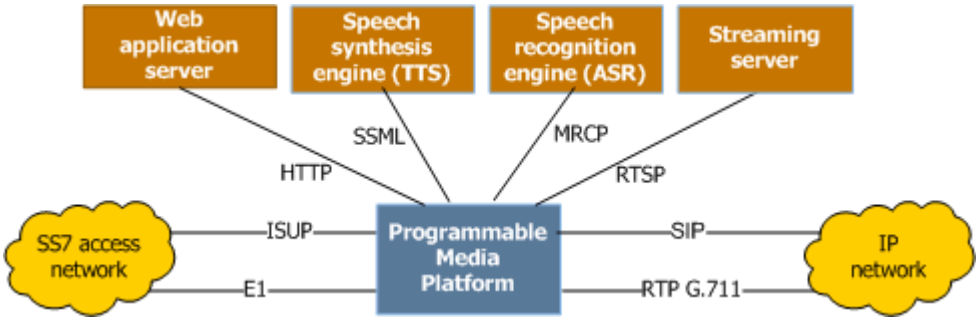
- Basic ISUP audio model
- Basic BICC audio model
- Basic ISUP video model
- Basic BICC video model
- ISUP scalable deployment model
- ISUP redundant deployment model

Basic ISUP audio model

In the basic ISUP audio model, the Programmable Media Platform:

- Provides ISUP signaling.
- Provides SIP signaling.
- Supports the Call Control Extensible Markup Language (CCXML) for call control.
- Executes VoiceXML applications and their associated media processing functions.

The following illustration shows the basic ISUP audio model:



Basic BICC audio model

In the basic BICC audio model, the Programmable Media Platform:

- Provides BICC signaling.

- Provides SIP signaling.
- Supports the Call Control Extensible Markup Language (CCXML) for call control.
- Executes VoiceXML applications and their associated media processing functions.

The following illustration shows the basic BICC audio model:

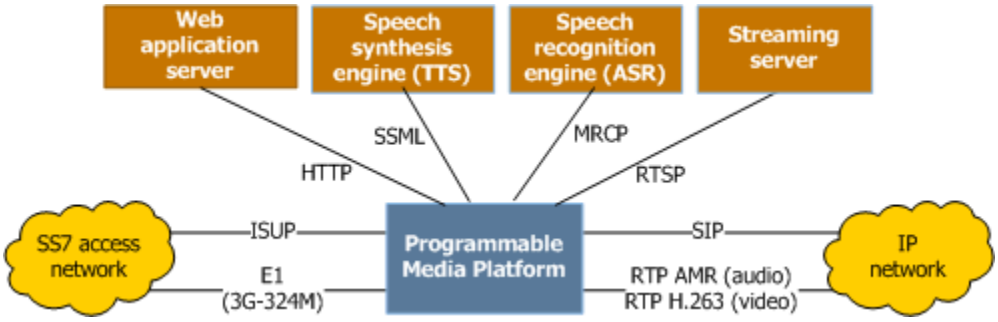
Basic ISUP video model

In the basic ISUP video model, the Programmable Media Platform:

- Provides ISUP signaling with 3G-324M.
- Provides SIP signaling.
- Supports the Call Control Extensible Markup Language (CCXML) for call control.
- Executes VoiceXML applications and their associated media processing functions.

The basic ISUP video model requires a mobile video device that supports 3G-324M.

The following illustration shows the basic ISUP video model:



Basic BICC video model

In the basic BICC video model, the Programmable Media Platform:

- Provides BICC signaling with 3G-324M.
- Provides SIP signaling.
- Supports the Call Control Extensible Markup Language (CCXML) for call control.
- Executes VoiceXML applications and their associated media processing functions.

The basic BICC video model requires a mobile video device that supports 3G-324M.

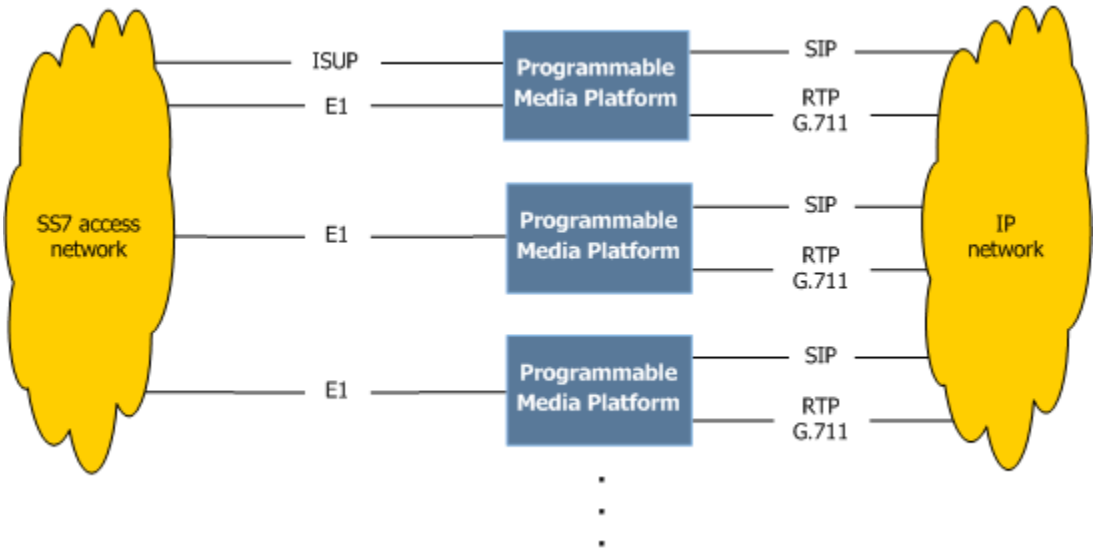
The following illustration shows the basic BICC video model:

ISUP scalable deployment model

For scalability, multiple Programmable Media Platforms can be deployed where the ISUP termination on one server provides the signaling between the SS7 access network and each of the other servers in the system.

In this model, the servers are configured to share the single signaling point code terminated by the server with ISUP interface. This configuration allows for high density deployments for a single signaling point code.

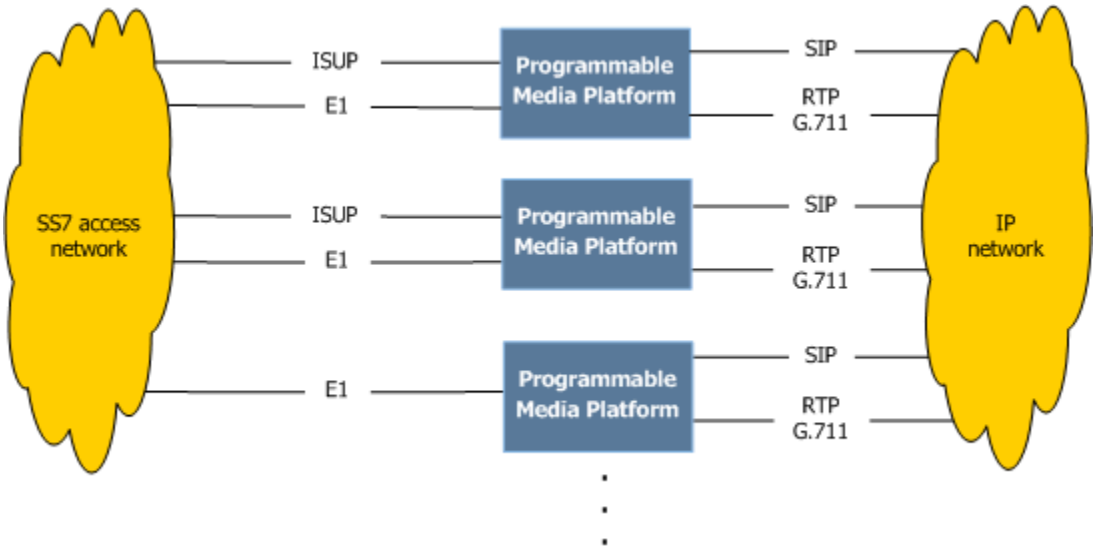
The following illustration shows the ISUP scalable deployment model.



ISUP redundant deployment model

For a redundant and fault-tolerant system, two Programmable Media Platforms can be deployed to provide higher availability. The servers share a single signaling point code and provide node-level redundancy.

The following illustration shows an ISUP model with two Programmable Media Platforms to support redundancy. The redundant pair of servers with ISUP terminations seamlessly provides signaling services for multiple servers as in the scalable deployment model.



IP models

The Programmable Media Platform is available with an IP-only audio model and an IP-only video model.

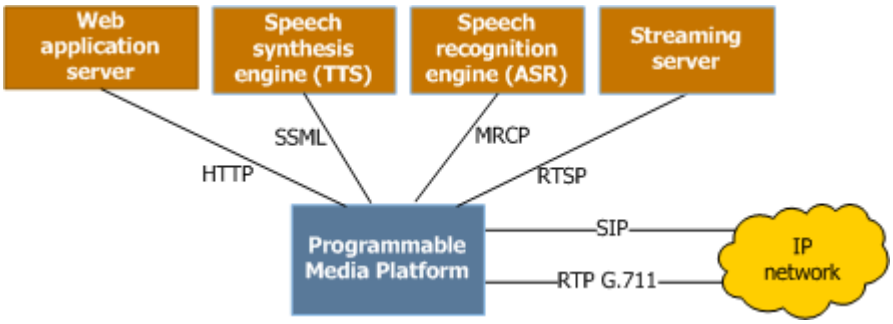
IP-only audio model

In the IP-only audio model, the Programmable Media Platform:

- Provides SIP signaling.
- Provides audio media capabilities.
- Executes VoiceXML applications and their associated media processing functions.

The web application uses standard VoiceXML operations to process audio. For information, see Developing an audio application and the *Dialogic® Vision™ VoiceXML Reference Manual*.

The following illustration shows the IP-only audio model:



Note: The IP-only audio model does not use media boards and does not include the Call Server.

IP-only video model

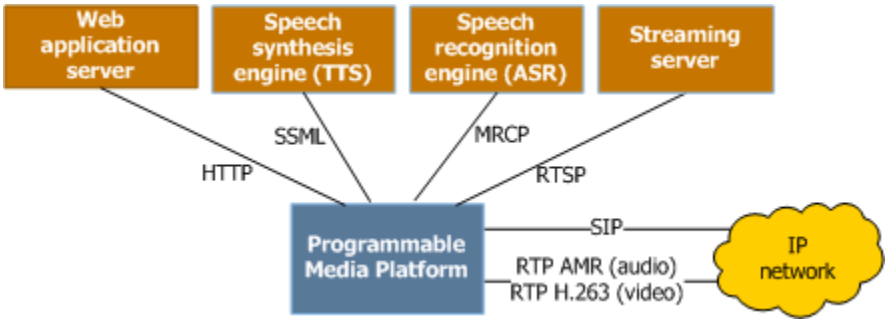
In the IP-only video model, the Programmable Media Platform:

- Provides SIP signaling.
- Provides audio capabilities.
- Provides video capabilities.
- Provides the option to transcode between AMR to G.711.
- Supports the Call Control Extensible Markup Language (CCXML) for call control.
- Executes VoiceXML applications and their associated media processing functions.

The web application uses standard VoiceXML operations to process audio, and extended VoiceXML operations to process video. It can play, record, and post video messages.

For more information, see Developing an audio application, Developing a video application, and the *Dialogic® Vision™ VoiceXML Reference Manual*.

The following illustration shows the IP-only video model:



Models with Video Transcoders

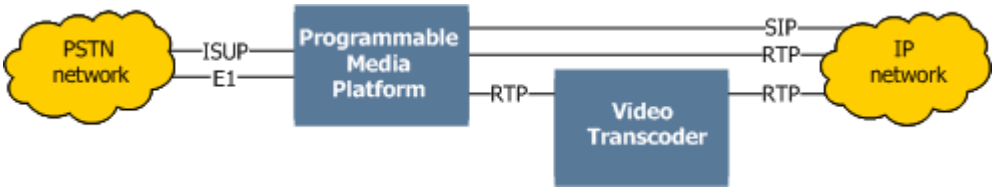
The Programmable Media Platform includes options for video transcoders. A video transcoder can be deployed in several configurations based on application need as described in the following topics.

- Video Transcoder interconnect
- Video model with a single Video Transcoder
- Video model with multiple Video Transcoders
- Video model with Programmable Media Platforms sharing Video Transcoders
- Video model with co-located Video Transcoder

For more information on video transcoders, see Managing video transcoder resources.

Video Transcoder interconnect

The Programmable Media Platform controls the video transcoder resources and inserts the transcoder in the video media path between the platform and the target IP endpoint. The communication interface to the IP endpoint is SIP. The interworking of the Programmable Media Platform has proprietary control of the video transcoder. Video traffic is routed to and from the video transcoder via RTP. Audio traffic flows separately from the Programmable Media Platform to the IP endpoint via RTP.

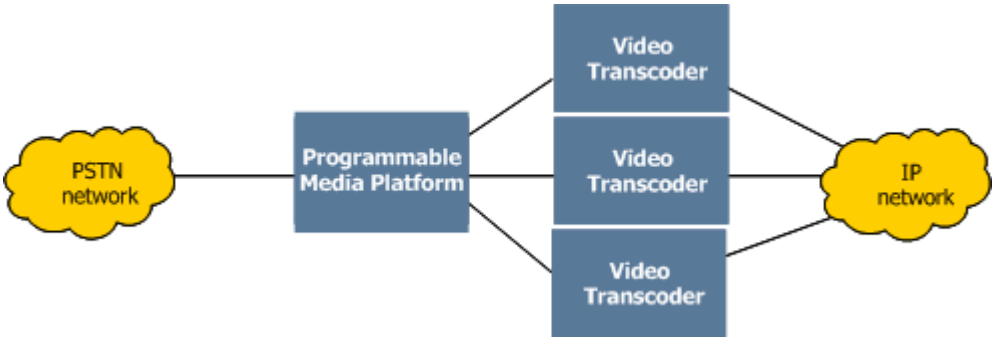


Video model with a single Video Transcoder

In a simple case, a video transcoder is mated with the Programmable Media Platform. The Programmable Media Platform is configured with this single video transcoder system which is used to complete Programmable Media Platform routes for the negotiated video codecs.

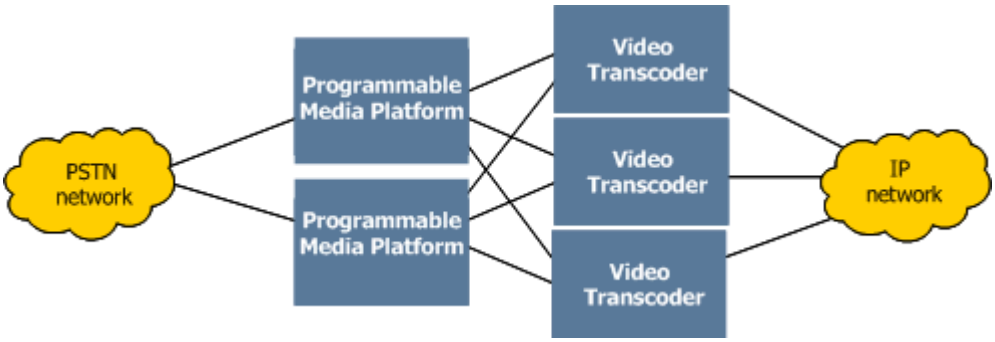
Video model with multiple Video Transcoders

For scalability in transcoding requirements, multiple video transcoder systems may be required to satisfy the needs of the application or the connectivity requirements of the target IP endpoints. For this reason, the Programmable Media Platform may be configured to use multiple video transcoders.



Video model with Programmable Media Platforms sharing Video Transcoders

For flexibility in deployments and scalability of a platform solution, multiple Programmable Media Platforms may be configured to share multiple video transcoder systems. In this configuration, the solution provider can view the composite of these servers as a single node or scalable Programmable Media Platform.



Video model with co-located Video Transcoder

The video transcoder may be a subsystem deployed on the same physical server as the Programmable Media Platform. Logically these are separate servers and are configured similarly to the Video model with a single Video Transcoder.

Standards

The Programmable Media Platform complies with and supports the following standards, depending on the model:

Standard	Version	Programmable Media Platform model
BICC	ITU-T Q.1901, 2000 ITU-T Q.1902-6, 2001 ANSI T1.673-2002[R2007]	All ISUP models
CCXML	Version 1.0, based upon the W3C Working Draft of CCXML dated 29th June 2005 See http://www.w3.org/TR/2005/WD-ccxml-20050629.	All models, except IP-only audio
ISUP	ITU White, ITU Blue, ITU 97, ETSI V2, ETSI V3, Q.767, ANSI 88, ANSI 92, ANSI 95, JNTT	All ISUP models
MRCP	A Media Resource Control Protocol Developed by Cisco, Nuance, and Speechworks, Shanmugham, Monaco, and Eberman, IETF Internet-Draft, draft-shanmugham-mrcp-05, January 2004.	All models
MTP	ITU, ANSI, JNTT, JTTC	All ISUP models
NETANN	Basic Network Media Services with SIP, IETF RFC 4240 See http://www.ietf.org/rfc/rfc4240.txt.	All models that use a SIP application server
OSP	Open Settlement Protocol ETSI TS 101 321	All models
RFC 2833	RTP Payload for DTMF Digits, Telephony Tones and Telephony Signal, Schulzrinne and Petrack, IETF RFC 2833, May 2000.	All models
RTSP	Real Time Streaming Protocol (RTSP), Schulzrinne, Rao, and Lanphier, IETF RFC 2326, April 1998.	All models

Standard	Version	Programmable Media Platform model
SIP	RFC 1889, RTP: A Transport Protocol for Real-Time Applications RFC 1890, RTP profiles RFC 2327, SDP: Session Description Protocol RFC 2833, RTP payload for DTMF digits RFC 3261, SIP: Session Initiation Protocol, Rosenberg et al., June 2002 RFC 3262, Reliability of Provisional Responses in SIP RFC 3263, Locating SIP servers RFC 3264, SDP Offer/Answer RFC 3311, SIP UPDATE method RFC 3325, Private Extensions to SIP for Asserted Identity within Trusted Networks RFC 3326, The Reason Header Field for SIP RFC 3398, ISDN ISUP to SIP mapping (partial support) RFC 3515, SIP Refer Method RFC 4040, RTP Payload Format for a 64 kbit/s Transparent Call RFC 4566, SDP: Session Description Protocol RFC 4694, Number Portability Parameters for the tel URI RFC 5009, Private Header Extension to SIP for Authorization of Early Media RFC 5168, XML Schema for Media Control (SIP VFU)	All models
SIP-VXML	SIP Interface to VoiceXML Media Services, Burke et al., IETF Internet-Draft, draft-burke-vxml-2, November 2006	All models
SISR	W3C Semantic Interpretation for Speech Recognition, Version 1.0	All models

Standard	Version	Programmable Media Platform model
SRGS	Speech Recognition Grammar Specification Version 1.0, Hunt et al, W3C Candidate Recommendation, 16th March 2004. See http://www.w3.org/TR/speech-grammar.	All models
SSML	Speech Synthesis Markup Language Version 1.0, Burnett et al, W3C Recommendation, 7th September 2004. See http://www.w3.org/TR/speech-synthesis.	All models
VoiceXML	Voice Extensible Markup Language (VoiceXML) Version 2.1, Oshry et al, W3C Candidate Recommendation, 13th June 2005. See http://www.w3.org/TR/voicexml21.	All models

Document conventions

By default, the Programmable Media Platform software is installed in the */opt/nms/vx* directory.

This manual uses the string *vx* to refer to the default installation directory.

Related documentation

The following manuals provide information related to installing and configuring the Programmable Media Platform:

Document	Description
Installing the Dialogic® Vision™ AQR1U Server	Describes how to install and cable the Dialogic® Vision™ AQR1U Server.
Installing the Dialogic® Vision™ Server TIGI2U	Describes how to install and cable the Dialogic® Vision™ Server TIGI2U.
Installing the Dialogic® Vision™ Server TIGW1U	Describes how to install and cable the Dialogic® Vision™ Server TIGW1U.
Dialogic® Vision™ VoiceXML Administration Manual	Describes how to configure the VoiceXML subsystem. Supplements the VoiceXML subsystem configuration information in this manual.

Document	Description
Dialogic® Vision™ Call Server Administration Manual	Describes how to configure the Call Server. Supplements the Call Server configuration information in this manual.
Dialogic® Vision™ Signaling Server Administration Manual	Describes how to configure the Signaling Server. Supplements the Signaling Server configuration information in this manual.
Dialogic® Vision™ CCXML Developer's Manual	Describes how to use the CCXML interface to configure and develop CCXML applications for the Programmable Media Platform.
Dialogic® Vision™ SNMP Reference Manual	Describes the management information bases (MIBs) and agents that support SNMP on the Programmable Media Platform.
Dialogic® Vision™ VoiceXML Reference Manual	An online manual that provides a VoiceXML language dictionary.
Dialogic® CG 6565 Media Board Installation and Developer's Manual	Describes how to configure the Dialogic® CG 6565 Series Media Boards.

3. Configuring the Dialogic® Vision™ 1000 Programmable Media Platform

Overview of configuring the Programmable Media Platform

All software is pre-installed and pre-configured on the Programmable Media Platform. However, software parameters are set for the manufacturing environment. You must re-configure some of these parameters so that the system operates properly at your site.

You must use the Dialogic® Vision™ Console to set up the Programmable Media Platform software. Using this web-based tool, you can enter field values and the tool automatically modifies the configuration files for your model.

Note: Attempting to generate a configuration manually may cause the configuration to be incompatible with the Vision Console and may render the Programmable Media Platform inoperable. For more information, see Avoiding conflicts with the Dialogic Vision Console.

This section describes how to use the Vision Console to set up the Programmable Media Platform software. It contains the following topics:

- Gathering information
- Logging into the Programmable Media Platform for the first time
- Accessing the Dialogic Vision Console
- Creating or revising a configuration
- Backing up a configuration
- Restoring a configuration
- Accessing the Programmable Media Platform using a secure shell
- Resetting the root password
- Installing a security certificate
- User account management
- Centralized user authentication

Gathering information

Before you configure the Programmable Media Platform, have the following types of information available:

- Network configuration information (all models)
- ISDN configuration information (ISDN models)
- Trunk configuration information (TDM models)
- Signaling server configuration information (ISUP models)
- ISUP configuration information (ISUP models)
- Video Transcoder configuration information (if applicable)
- IP-324M configuration information (if applicable)
- Ethernet redundancy configuration information (if applicable)
- Network monitor configuration information (if applicable)

- Node configuration information (if applicable)
- SIP load balancing configuration information (if applicable)

Network configuration information

The following information is required for all Programmable Media Platform models, except where indicated:

Required information	Value
Domain name for the Programmable Media Platform	
DNS server IP addresses	
Programmable Media Platform Ethernet 0 IP address, subnet mask, and default server	
Programmable Media Platform Ethernet 1 IP address, subnet mask, and default server	
(All models except IP-only audio) Media board 0 IP address, subnet mask, and default server	
(All models that have two media boards) Media board 1 IP address, subnet mask, and default server, if present.	
IP address of the application server where the VoiceXML index is hosted	
Port of the HTTP server for the application server where the VoiceXML index is hosted	

ISDN configuration information

If you are using the ISDN audio or ISDN video model, gather the following ISDN configuration information:

Required information	Value
ISDN protocol variant	
ISDN equipment type	

Trunk configuration information

The following information is required for all Programmable Media Platform models, except IP-only audio and IP-only video:

Required information	Value
CRC	☐ On ☐ Off
Line impedance	☐ DSX1 ☐ G703_75_OHM ☐ G703_120_OHM
Frame type	☐ E☐F ☐ D4 ☐ CEPT
Line coding	☐ AMI ☐ B8ZS (T1) ☐ HDB3 (E1) ☐ AMI_ZCS (T1) ☐ AMI_BELL (T1) ☐ AMI_DS (T1) ☐ AMI_GTE (T1)
Label or identification used to physically identify each E1 trunk on media board 0	Voice trunks (can also carry signaling): Trunk 1: _____ Trunk 2: _____ Trunk 3: _____ Trunk 4: _____ Trunk 5: _____ Trunk 6: _____ Trunk 7: _____ Trunk 8: _____ Signaling only trunks: Trunk 9: _____ Trunk 10 _____

Required information	Value
(For implementations with two media boards) Label or identification used to physically identify each E1 trunk on media board 1	Voice trunks (can also carry signaling): Trunk 1: _____ Trunk 2: _____ Trunk 3: _____ Trunk 4: _____ Trunk 5: _____ Trunk 6: _____ Trunk 7: _____ Trunk 8: _____ Signaling only trunks: Trunk 9: _____ Trunk 10 _____

Signaling server configuration information

The following information is required for ISUP models that have a Signaling Server:

Required information	Value
Signaling server IP address	
Signaling server circuit start value	
Signaling server variant (switch type)	☐ ANSI88 ☐ ANSI92 ☐ ANSI95 ☐ ANSIBICC ☐ ETSIV2 ☐ ETSIV3 ☐ ITU97 ☐ ITUBICC ☐ ITUBLUE ☐ ITUWHITE ☐ JNTT ☐ Q767

Required information	Value
Name of the signaling server associated with each trunk. Values for CG media board 1 are used for implementations with multiple media boards.	Media board 0: Trunk 1: _____ Trunk 2: _____ Trunk 3: _____ Trunk 4: _____ Trunk 5: _____ Trunk 6: _____ Trunk 7: _____ Trunk 8: _____ Media board 1: Trunk 1: _____ Trunk 2: _____ Trunk 3: _____ Trunk 4: _____ Trunk 5: _____ Trunk 6: _____ Trunk 7: _____ Trunk 8: _____

ISUP configuration information

If you are using the basic ISUP audio or basic ISUP video model, gather the following configuration information:

- MTP/M3UA common information
- MTP 1 information (only applicable if MTP transport is required)
- MTP 2 and MTP 3 information (only applicable if MTP transport is required)
- M3UA information (only applicable if SIGTRAN transport is required)
- Peer signaling process information (only applicable if SIGTRAN transport is required)
- ISUP/BICC information

MTP/M3UA common information

Required information	Value
Transport	☐ MTP ☐ SIGTRAN

Required information	Value
PC format	☐ 3.8.3 (14 bits) ☐ 8.8.8 (24 bits) Other _____
Local point code	

MTP 1 information (only applicable if MTP transport is required)

Required information	Value
Number of E1s/T1s required	T1 _____ E1 _____
How E1s are being presented	BNC Male (75 ohms) RJ48 (120 ohms) RJ45 (120 ohms)
Line coding	AMI B8ZS (T1) HDB3 (E1) AMI_ZCS (T1) AMI_BELL (T1) AMI_DS (T1) AMI_GTE (T1)
Frame type	ESF D4 CEPT
CRC	On Off

Required information	Value
Label or identification used to physically identify each E1 trunk	Voice trunks (can also carry signaling): Trunk 1: _____ Trunk 2: _____ Trunk 3: _____ Trunk 4: _____ Trunk 5: _____ Trunk 6: _____ Trunk 7: _____ Trunk 8: _____ Signaling only trunks: Trunk 9: _____ Trunk 10 _____

MTP 2 and MTP 3 information (only applicable if MTP transport is required)

Required information	Value
Number of links	
SS7 variant	
Trunk number for link 1	
Timeslot for link 1	
Adjacent point code for link 1	
Signaling link code (SLC) for link 1	
Subservice field link for link 1	
Trunk number for link 2	
Timeslot for link 2	
Adjacent point code for link 2	
Signaling link code (SLC) for link 2	
Subservice field link for link 2	

M3UA information (only applicable if SIGTRAN transport is required)

Required information	Value
Local routing context	
Network appearance code	
Service variant	☐ ANSI ☐ BICC ☐ ITU ☐ CHINA ☐ NTT ☐ TTC
SCTP source port	

Peer signaling process information (only applicable if SIGTRAN transport is required)

Required information	Value
Destination IP address	
SCTP port	
Peer type	☐ IPSP ☐ SGP
IPSP mode	☐ DE ☐ SE
Dynamic routing key management	
Use network appearance	
Client side	

ISUP/BICC information

Required information	Value
Origination point code	
Subservice field	

Required information	Value
Destination point codes for circuits on each T1/E1 trunk	
Number of circuits used per trunk	
Circuit identification code (CIC) for each T1/E1 trunk	
Trunk direction for each T1/E1 trunk	
ISUP variant	
Is inbound call required?	
Is outbound call required?	
Is transfer required? If yes, what type?	
Range of numbers to use for the Media Platform	

Video Transcoder configuration information

The following information is required for models that use video transcoding:

Required information	Value
IP address of video transcoder system	
IP address of second video transcoder system, if used	
IP address of third video transcoder system, if used	
IP address of *n* video transcoder system, if used	

IP-324M configuration information

The following information is required for models that support 3G-324M calls over IP:

Required information	Value
IP-324M support	☐ Enabled

Ethernet redundancy configuration information

If you plan to use the Ethernet redundancy feature, gather the following information as applicable:

- Network information
- SIP network
- RTP network
- Circuit-switched signaling network
- NbUP network
- Billing network
- OA&M network
- Signaling redundant network
- Routes configuration information

For information on Ethernet redundancy, see Working with Ethernet redundancy.

Network information

Determine the network addresses for each of the separate networks you need to address. Specify a VLAN ID if you require VLAN tagging for the traffic on these networks.

Network name	Network address	VLAN ID
SIP		
RTP		
Circuit-switched signaling		
NbUP		
OA&M		
Billing		
Signaling redundancy		

SIP network

Determine the IP address information related to your SIP network for each component of the server.

For configuration instructions, see Configuring the SIP network.

Device	IP address	Subnet mask	Default gateway
Server host			

RTP network

If you do not use a separate network for RTP traffic, disregard this section.

Determine the IP address information related to your RTP network for each component of the server.

For configuration instructions, see Configuring the RTP network.

Device	IP address	Subnet mask	Default gateway
Server host			
Media board #1			
Media board #2			

Circuit-switched signaling network

If you do not require SIGTRAN or BICC traffic in your system, disregard this section.

Determine the IP address information related to your circuit-switched signaling network for each component of the server.

For configuration instructions, see Configuring the Circuit-Switched Signaling network.

Device	IP address	Subnet mask	Default gateway
Signaling board			

NbUP network

If you do not use a separate network for NbUP traffic, disregard this section.

Determine the IP address information related to your NbUP network for each component of the server.

For configuration instructions, see Configuring the NbUP network.

Device	IP address	Subnet mask	Default gateway
Media board #1			
Media board #2			

Billing network

If you do not use a separate network for Billing traffic, disregard this section.

Determine the IP address information related to your Billing network for each component of the server.

For configuration instructions, see Configuring the Billing network.

Device	IP address	Subnet mask	Default gateway
Server host			

OA&M network

If you do not use a separate network for operations, administration, and management, disregard this section.

Determine the IP address information related to your OA&M network for each component of the server.

For configuration instructions, see Configuring the OA&M network.

Device	IP address	Subnet mask	Default gateway
Server host			

Signaling Redundant network

If you do not use redundant servers to implement circuit-switched signaling redundancy, disregard this section.

Determine the IP address information related to your signaling redundant network for each component of the server.

For configuration instructions, see Configuring the Signaling Redundant network.

Device	IP address	Subnet mask	Default gateway
Signaling board			

Routes configuration information

If you plan to use the network redundancy feature, gather the following routes information.

Host routes

Network type	Network IP address	Subnet mask	Primary destination	Backup destination
SIP				
RTP				
OA&M				
Billing				

Media boards routes

Network type	Network IP address	Subnet mask	Primary destination	Backup destination
RTP				
NbUP				

For more information, see Network redundancy and the network monitor service.

Network monitor configuration information

If you plan to use the network monitor service, specify IP addresses to be monitored for each network.

Network name	Primary address(es)	Backup address(es)	Monitored from
SIP			Host
RTP			Host and Media boards
OA&M			Host

Network name	Primary address(es)	Backup address(es)	Monitored from
Billing			Host
NbUP			Media boards

For more information, see Network redundancy and the network monitor service.

Node configuration information

If you intend to group Vision Servers into a Vision node, specify the node name and the IP address for each node member.

The node member name is automatically derived from the node name by appending a dash and a sequential number to the node. For example, if the node name is VISION, the node members will be named VISION-1, VISION-2, and so on.

Node name	Node member IP address
	Member 1:
	Member 2:
	Member 3:
	Member 4:
	Member 5:
	Member 6:
	Member 7:
	Member 8:

For more information, see Managing Vision Nodes.

SIP load balancing configuration information

If you intend to use SIP load balancing, you must choose a virtual IP address for the single SIP entry point. The virtual IP address must be on the same network as the signaling network.

For more information on this feature, see Using SIP load balancing.

Logging into the Programmable Media Platform for the first time

The information in this topic assumes you have installed and cabled the Programmable Media Platform, as described in the installation manuals (see Related documentation).

The Programmable Media Platform is shipped from the manufacturer with the following default IP network configuration for the first Ethernet interface (eth0):

- IP address: 192.168.0.1
- Subnet mask: 255.255.255.0
- Gateway: None
- Host name: VISION

You must use the Vision Console to configure the IP address for the Programmable Media Platform. You can configure the Programmable Media Platform to use a static IP address (recommended) or DHCP.

Configuring the Programmable Media Platform to use a static IP address

To configure the Programmable Media Platform to use a static IP address, follow these steps:

Step	Action
1	Assign IP address 192.168.0.100 to the computer that will access the Dialogic® Vision™ Console.
2	Connect the Vision Console computer to eth0 on the Programmable Media Platform either directly using a crossover cable, or connect through a standalone Ethernet hub or switch.
3	Enter the following URL from an Internet Explorer or Firefox browser on the Vision Console computer: http://192.168.0.1 You are redirected to secure HTTP (HTTPS) connection. A message related to the web site's security is displayed. You can click to continue to the web site, or you can install a security certificate on the system. For information, see Installing a security certificate. For terminal or command line access, you may use secure shell (SSH) or optionally connect via KVM (but KVM is not recommended for deployed systems).
4	Log into the Vision Console using the following information: • User: vision-sys-admin • Password: Vision_1000 The Overview page appears.

Step	Action
5	Click **Host IP information** in the Configuration menu. The Host IP information page appears.
6	Change the value of the IP address field for eth0 to the desired IP address. Then set appropriate values for the Subnet mask and Gateway fields.
7	Configure the server's unique hostname, and then enter DNS server addresses in the DNS servers section, if required.
8	Double check the host IP information, and click **Submit** to apply the changes. The Vision Console attempts to reconnect after 15 seconds. If the server is on a different IP subnet than the Vision Console computer, the connection fails.
9	To validate that the gateway IP address is correct, change the IP address of the Vision Console machine to match the IP address subnet of the Programmable Media Platform. Then access the Vision Console by entering the IP address used in Step 6.

Obtaining an IP address through DHCP

If you need to use DHCP for the platform IP address, follow these steps:

Step	Action
1	Set up the DHCP server so the hostname and IP address assigned to the Programmable Media Platform are predetermined, for example, based on the platform's MAC address.
2	Assign IP address 192.168.0.100 to the computer that will access the Dialogic® Vision™ Console.
3	Connect the Vision Console computer to eth0 on the Programmable Media Platform either directly using a crossover cable, or through a standalone Ethernet hub.
4	Enter the following URL from an Internet Explorer or Firefox browser on the Vision Console computer: http://192.168.0.1 You are redirected to secure HTTP (HTTPS) connection. A message related to the web site's security is displayed. You can click to continue to the web site, or you can install a security certificate on the system. For information, see Installing a security certificate. For terminal or command line access, you may use secure shell (SSH) or optionally connect via KVM (but KVM is not recommended for deployed systems).

Step	Action
5	Log into the Programmable Media Platform using the following information: • User: vision-sys-admin • Password: Vision_1000 The Overview page appears.
6	Click **Host IP information** in the Configuration menu. The Host IP information page appears.
7	Click **DHCP** next to eth0 to enable DHCP.
8	Click **Submit**.
9	Connect the Programmable Media Platform to the network.
10	Once the Programmable Media Platform has acquired its IP address and is reachable through a ping, access the Vision Console.
11	Click **Services** on the Operations menu.
12	Click **Reboot**, and wait for the system to restart before continuing.

Accessing the Dialogic Vision Console

Use the Dialogic® Vision™ Console to configure and manage the Programmable Media Platform or a Vision node. To access the Vision Console, follow these steps:

Step	Action
1	Open one of the following local browsers. It is helpful to view the pages in full screen mode. • Firefox 1.0 or later • Microsoft® Internet Explorer® 6.0 or later
2	If this is the first time you are accessing the Vision Console, you must set up a host IP address for it as described in Logging into the Programmable Media Platform for the first time. Once you have configured the host IP address for the Vision Console, enter the configured address.

Step	Action
3	Enter the following information: • User name: (a user name listed below) • Password: Vision_1000 The default password for all user names is Vision_1000. The Vision Console displays the Overview page, which contains information about the servers, media boards, port rating, software version, installed patches, and licensing information for the Programmable Media Platform. This page also shows whether conferencing is enabled.

The user names for the Vision Console have the following rights:

User name	Associated rights
vision-root	Super-user. This user can do everything a system administrator can do. In addition, this user can change the Programmable Media Platform model with assistance from Dialogic Technical Services and Support. This user can also delete log files and CDR files. **Note:** Appropriate licenses are required to change the Programmable Media Platform model.
vision-sys-admin	System administrator. This user can do everything an application administrator can do. In addition, this user can change System menu settings, Configuration and Provisioning menu settings, and can perform a port capacity upgrade. **Note:** Appropriate licenses are required to perform a port capacity upgrade.
vision-app-admin	Application administrator. This user can monitor and add routes, and monitor and add CCXML applications. The options in the Configuration and Operations menus are read-only for this user.
vision-guest	End user. This user has read-only access in all menus.

For more information, see User account management.

Creating or revising a configuration

This topic describes how to create or revise a configuration for a Programmable Media Platform or a Vision node using the Dialogic® Vision™ Console. For information about configuration parameters, see the Vision Console parameters section.

Create a configuration

To create a configuration using the Vision Console, follow these steps:

Step	Action
1	Access the Vision Console, and log in as vision-sys-admin. The Overview page appears with information about the version of the Programmable Media Platform.
2	If you intend to group multiple Vision Servers into a Vision node, define the node. Click **Node Definition** on the Configuration menu. See Managing Vision Nodes for more information.
3	Check the host IP information settings. These values are set during initial setup, as described in Logging into the Programmable Media Platform for the first time. Click **Host IP information** on the Configuration menu, and fill in the fields. Click **Submit**.
4	Click **Resources** on the Configuration menu, and fill in the fields. Click **Submit**.
5	Click **SIP** on the Configuration menu, and fill in the fields. Click **Submit**.
6	Click **RTP** on the Configuration menu, and fill in the fields. Click **Submit**.
7	Enter PSTN parameters, if you are configuring an ISDN audio, ISDN video, or ISUP video model; or an ISUP audio model with signaling server integration. Click **PSTN** on the Configuration menu, and fill in the fields. Click **Submit**.
8	Click **Services** on the Operations menu, and then click **Restart all**. Once the status of all services is STARTED, you can proceed.
9	Check the status of all services from the Monitoring menu; these services should be online.
10	Depending on the server model you are configuring, you might also need to manually specify additional configuration settings, as described in Additional configuration tasks.

Revise a configuration

To revise a configuration using the Vision Console, follow these steps:

Step	Action
1	Access the Vision Console, and log in as vision-sys-admin. The Overview page appears with information about the version of the Programmable Media Platform.
2	Make the appropriate parameter changes.
3	If you changed any Configuration menu parameters, restart all services. Click **Services** on the Operations menu, and then click **Restart all**. Once the status of all services is STARTED, you can proceed.

Additional configuration tasks

The following table lists additional configuration tasks you may need to perform after using the Vision Console, depending on the configured environment:

Models	Configuration task	For more information, see...
All	Change H.100 clocking configuration.	Fine tuning the H.100 clocking configuration.
ISUP audio and ISUP video models	Add additional settings for SS7 signaling trunk in the */opt/nmstx/etc/cx/txcfg1.txt* file.	
	Add TX IP information required for SIGTRAN in */opt/nmstx/etc/cx/ipcfg1.txt*.	
	Add MTP3, SIGTRAN, and ISUP configuration information in */opt/hs-data/raid/nms_hearsay/cfg/oam/ss7_config_default.xml*.	*Dialogic® Vision™ Signaling Server Administration Manual.*

Backing up a configuration

To back up an existing Programmable Media Platform or Vision node configuration, follow these steps:

Step	Action
1	Access the Vision Console, and log in as vision-sys-admin.
2	Click **Import/Export** on the Configuration menu. The Import/Export configuration page appears.

Step	Action
3	Under Export current configuration, click **Save As**. The File Download window appears.
4	Click **Save**, locate the directory where you want to store the downloaded configuration, and enter the file name in the File name field. The configuration is downloaded to a *.zip* file.
5	Click **Save**. The system backs up the configuration and displays a message.
6	Click **OK**.

Restoring a configuration

To restore a Programmable Media Platform or Vision node configuration, follow these steps:

Step	Action
1	Access the Vision Console, and log in as vision-sys-admin.
2	Click **Import/Export** in the Configuration menu. The Import/Export page appears.
3	Under Import configuration, click **Browse**, and locate the configuration you want to restore.
4	Select the configuration elements to be restored, such as base configuration or network configuration. For a Vision node, you can choose to restore the configuration of the full node or specific node members.
5	Click **Apply**. A confirmation message displays.
6	Click **OK**. The system restores the configuration.
7	Click **Refresh** to display the Overview page.
8	Click **Services** on the Operations menu, and then click **Restart all** to restart the Programmable Media Platform services.

Accessing the Programmable Media Platform using a secure shell

The Programmable Media Platform is normally managed through the Vision Console. However, in some situations (for example, to take actions requested by Dialogic Technical Services and Support), you may need to use a secure shell (ssh) to log into the Programmable Media Platform.

Any user name defined in the database can access the Programmable Media Platform with ssh, but only users with super-user rights can run root commands. You must run root commands using the sudo command.

To use ssh to log into the Programmable Media Platform, follow these steps:

1. Use an ssh client to establish a secure shell connection.
   For example: ssh vision-sys-admin@192.168.0.1
   Replace 192.168.0.1 with the IP address of your Programmable Media Platform.
2. When prompted, enter the user password (Vision_1000 is the default password).
   You now have access to a standard bash shell.

To run root commands, follow these steps:

1. Connect to the Programmable Media Platform using a user name with super-user rights.
   For example: ssh vision-root@192.168.0.1
2. Use sudo to run the command that requires root privileges.
   For example:
   sudo ifconfig
   sudo/bin/bash

Resetting the root password

If you can no longer connect to the Vision Server using one of the user names defined in the database, you can change the root password on the Vision Server by booting the server in single-user mode.

To change the root password on the Vision Server, follow these steps:

1. Connect a keyboard and monitor to the Vision Server.
2. Boot the Vision Server.
3. At the boot loader boot screen, select the kernel and press **e**.
4. Select the second line (the line starting with the word kernel) and press **e**.
5. Append the word **single** to the end of the line and press **Enter**.
6. Press **b** to boot the kernel.
7. Once the kernel is booted, enter the **passwd** command followed by the new password.
8. Reboot the server by entering the **reboot** command.

Installing a security certificate

The Dialogic® Vision™ Server provides secure HTTP (HTTPS) access. It uses a self-signed certificate, which means that it is generated by the server itself and not by a known certificate authority. This self-signed certificate does not present a security risk.

When you log into the server for the first time, a message about the web site's security certificate or a message about untrusted connection is displayed, depending on the browser.

You can click to continue to the web site, or you can install a security certificate on the system so that the security message won't be displayed each time you log in.

Note: These steps may differ depending on the browser version you are using.

On the Firefox browser, follow these steps to create a trusted connection:

Step	Action
1	After you enter the URL for the gateway in the browser, the message *This Connection is Untrusted* is displayed. Click **I Understand the Risks**. Information about the risks is displayed.
2	Click **Add Exception**. The Add Security Exception dialog box is displayed.
3	Ensure that the check box for **Permanently store this exception** is checked.
4	Optionally click **View** to verify the information about the certificate.
5	Click **Confirm Security Exception** to complete the process for creating a trusted connection.
6	Once installed, the trusted connection expires after one year. The trusted connection also expires if you upgrade the Dialogic® Vision™ Server software. In these cases, repeat Steps 1-5 to recreate a trusted connection.

On the Microsoft® Internet Explorer® browser, follow these steps to install a security certificate:

Step	Action
1	After you enter the URL for the gateway in the browser, the message *There is a problem with this website's security certificate* is displayed. Click **Continue to this website**. The Vision Console main page is displayed.
2	Next to the URL drop-down list, click **Certificate Error**. The Untrusted Certificate dialog box is displayed.
3	Click **View certificates** to view information about the certificate. The Certificate Information dialog box is displayed.
4	Click **Install Certificate**. The Certificate Import Wizard dialog box is displayed.

Step	Action
5	Accept the default responses and click **Next** until you reach the final question; then click **Finish**. A security warning message is displayed.
6	Click **Yes** to complete the process for installing the certificate.
7	Once installed, the certificate expires after one year. The certificate also expires if you upgrade the Dialogic® Vision™ Server software. In these cases, repeat Steps 1-6 to reinstall the security certificate.

User account management

The system administrator can add and modify user accounts, set and reset passwords, specify a user's access level, and more through user account management.

Default user names are provided by the Vision Server, as described in Accessing the Dialogic Vision Console. User account information is stored in a single database and is managed through the Vision Console.

For related information, see Centralized user authentication.

Creating a new user account

Follow these instructions to create a new user account:

1. Access the Vision Console as described in Accessing the Dialogic Vision Console, and log in with an account that has system administrator rights, such as vision-sys-admin.
2. Click **User administration** on the System menu. The User administration page is displayed.
3. Click **Add** and fill in the fields for the new user, including a unique user name, password, and access level. Then click **Submit.**
4. Repeat step 3 for each new user account that you wish to create.

Modifying a user account

Follow these instructions to modify a user account:

1. Access the Vision Console as described in Accessing the Dialogic Vision Console, and log in with an account that has system administrator rights, such as vision-sys-admin.
2. Click **User administration** on the System menu. The User administration page is displayed.
3. Click **Edit** next to the user name that you wish to modify. The properties of this user name are displayed.
4. Edit the properties as required and click **Submit**.
5. Repeat steps 3-4 for each user account that you wish to modify.

Removing a user account

Follow these instructions to remove a user account:

Note: Default user accounts provided by the Vision Server may be edited but may not be deleted.

1. Access the Vision Console as described in Accessing the Dialogic Vision Console, and log in with an account that has system administrator rights, such as vision-sys-admin.
2. Click **User administration** on the System menu. The User administration page is displayed.
3. Click **Remove** next to the user name that you wish to delete.
4. Repeat step 3 for each user name that you wish to delete.

Centralized user authentication

The Vision Server allows the creation of a centralized user database which permits user name and password information to be shared among multiple servers.

Authentication information is managed by a Lightweight Directory Access Protocol (LDAP) server. On standalone servers, only the local LDAP server is referenced. When a centralized authentication database is shared among multiple Vision Servers, authentication requests are sent over the network to the acting LDAP server. These message exchanges are encrypted and require the use of a server certificate.

For information about creating user accounts, see User account management.

Types of LDAP servers

A **Provider server** is the master server. All updates to the database which contains the user information are made through the Provider server.

A **Consumer server** is a slave to the Provider server. Consumer servers are notified of changes to the Provider server database when they occur; for example, a new user is added or a user is removed.

Consumer servers are used for replication. Consumer servers can function even when the Provider server is offline. If the Provider server is down, users can still log in using a Consumer server.

If your environment uses multiple Vision Servers, you can configure one server as a Provider and all other servers as a Consumer. This set up allows you to use the same user names and passwords on each server.

A **Standalone** server is one in which the server acts as an LDAP master server and is only accessed by the local server.

By default, the Programmable Media Platform is configured as a Standalone server.

Configuring the Provider server

Follow these instructions to configure the Provider server:

1. Access the Vision Console as described in Accessing the Dialogic Vision Console, and log in as vision-sys-admin.
2. Click **Authentication** on the System menu. The User authentication page is displayed. By default, the Programmable Media Platform is configured as a Provider

server with read-only database access. For a description of the fields, see System menu parameters.

3. Configure fields on this page as required and click **Submit**.

Configuring the Consumer server

You can view or configure the following information on a Consumer server:

- View and configure the IP address of the Provider server.
- View the status of the Provider server certificate.
- Pull and install the Provider server certificate.

Follow these instructions to configure the server type as Consumer:

Note: This procedure assumes that you have previously configured user authentication settings on the Provider server.

1. Access the Vision Console as described in Accessing the Dialogic Vision Console, and log in as vision-sys-admin.
2. Click **Authentication** on the System menu. The User authentication page is displayed. By default, the Programmable Media Platform is configured as a Provider server with read-only database access. For a description of the fields, see System menu parameters.
3. Under LDAP server setting, select **Consumer** as the type.
4. Under Provider, enter the IP address and port of the Provider server.
5. Click **Install** to install the Provider server certificate on the Consumer server. The server status is updated. For example, the status can be Trusted, Untrusted, or Unavailable.

4. Dialogic® Vision™ Console parameters

Configuration menu parameters

The Configuration menu contains the following pages:

- Overview
- Node definition
- Date and Time
- Host IP information
- Resource configuration
- SIP parameters
- RTP parameters
- NbUP circuits
- Trunks
- PSTN
- Signaling Server
- Options
- Capacity upgrade
- SNMP configuration
- Network redundancy configuration
- Video Transcoder
- Import/Export configuration

Note: If your environment includes multiple Vision Servers defined as a Vision node, some pages (such as Date and Time, Options, SNMP) contain information that is common to all node members, while other pages (such as Host IP information, Resources) contain server-specific information. Use the node navigation menu to select and submit the configuration for each node member.

Overview

The Overview page displays information about the current Programmable Media Platform configuration such as:

- Model type and version
- Media board information
- Port rating
- Whether conferencing is enabled
- License information

Node definition

The Node definition page allows you to group two or more Vision Servers in a Vision node. Access this page by clicking **Node definition** on the Configuration menu.

Group	Parameter	Description
Node	Node name	Node name, such as VISION.
Node members	Member name	Name of member that belongs to this node. This name is automatically derived from the node name by appending a dash and a sequential number to the node. For example, if the node name is VISION, the node members will be named VISION-1, VISION-2, and so on.
	IP address	IP address of the node member.
	Enabled	Click the check box to enable this node member. Clear the check box to disable the node member.
	Edit	Click **Remove** to remove a node member. Node members will automatically be renamed to be sequential as necessary. Click **Update** to update node member information. Click **Discover** to view a list of Vision Servers on the present Ethernet segment that are not already part of a node.
	Add	Click **Add** to add a node member.
	Deploy	Click **Deploy** to apply the node definition to the node members, assuming that all members are available.

For more information, see Managing Vision Nodes.

Date and Time

The Date and Time page configures date and time settings for the Programmable Media Platform. Access this page by clicking **Date and Time** on the Configuration menu.

Note: Before changing the date and time settings, you should stop services from the Services page of the Operations menu.

Parameter	Description
Date	Date. Example: Wed Nov 11 2009. Click the calendar icon to change the date.
Time zone	Time zone. Example: America/Montreal. Click the arrow in the drop-down list to change the time zone.
Time	Time. Click the up or down arrow to change the hour or minutes.

Parameter	Description
NTP	Network Time Protocol. Use NTP to synchronize time and date across multiple servers. Click the check box to enable Network Time Protocol. If enabled, the NTP Server #1 and NTP Server #2 parameters are displayed.
NTP Server #1	IP address of NTP Server #1, if used.
NTP Server #2	IP address of NTP Server #2, if used.

Host IP information

The Host IP information page configures the IP network settings for the Programmable Media Platform. Access this page by clicking **Host IP information** on the Configuration menu.

For a Vision node, this page displays server-specific information. Use the node navigation menu to select and configure a node.

Group	Parameter	Description
Interface configuration	Interface	Ethernet interface, such as eth0 and eth1, or interface alias, such as eth0:1 and eth1:1.
	Enabled	Indicates whether the interface is active and whether two interfaces are bonded. Bonded interfaces share the same bond device value. For example, to bond eth0 and eth1, set this parameter to bond0 for both interfaces.
	DHCP	Indicates whether DHCP is enabled for the interface.
	IP address	IP address for the interface, if DHCP is not enabled.
	Subnet mask	Subnet mask for the interface, if DHCP is not enabled.
	Gateway	IP address of the default gateway for the Programmable Media Platform network card, if DHCP is not enabled.
	VLAN	Virtual LAN (VLAN) ID, used to enable on-host VLAN tagging.

Group	Parameter	Description
	VIP	Virtual IP address, used for SIP load balancing.
	Edit	To create an alias for the selected interface which may be a physical Ethernet port or a bond device, click **Add alias**. A new entry is added at the bottom of the table. To remove an interface definition, click **Remove**.
Traffic types	Type	Type of traffic for on-host services. • Signaling - interface that handles SIP traffic • Media - interface that handles RTP traffic • OA&M - interface that handles administration and management tasks such as SNMP • Billing - interface that handles Billing traffic
	Interface	Interface associated with the traffic type. If set to **any** for Media, OA&M, and Billing, this means that the traffic is not bound to any specific interface. If set to **any** for Signaling, this means that the traffic is bound to the first interface discovered in the system in this order: 1. Native interfaces followed by their non-VLAN aliases. 2. VLAN-enabled native interfaces in order of VLAN IDs. 3. Bonding interfaces followed by their non-VLAN aliases. 4. VLAN-enabled bonding interfaces in order of VLAN IDs.
Hostname	Hostname	Host name for the Programmable Media Platform.
DNS Servers	Server #1 - Server #3	IP address of the domain name servers for the Programmable Media Platform.
IP routes	IP address	IP address for the IP route of a configured interface. Routes cannot be assigned to alias interfaces. You can assign the route to the parent of the alias; the operating system will route packets to the appropriate alias. If the alias is VLAN-enabled, then you can assign a route directly to it.

Group	Parameter	Description
	Subnet mask	Subnet mask for the IP route of a configured interface.
	Destination	Destination IP address for the IP route of a configured interface.
	Backup destination	Backup destination IP address for the IP route of a configured interface. Used when the network monitor service performs a failover or a switchover to the backup network.
	Interface	Interface associated with the IP route.
	Edit	To remove an IP route definition, click **Remove**.
	Add	To create an IP route for a configured interface, click **Add**. Use the arrows below the table to reorder routes as needed.

Resource configuration

The Resource configuration page configures and enables functionality such as conferencing, T.38 fax, and video transcoding. It also configures the size of codec and conferencing resource pools for the media boards in the Programmable Media Platform.

Access the Resource configuration page by clicking **Resources** on the Configuration menu. The information on this page varies with the configuration, such as Programmable Media Platform with a Video Transcoder system. The Resource configuration page automatically opens in basic mode. By default, all codecs support RFC 2833 encoding, decoding, and DTMF detection.

In addition, for a Vision node, this page displays server-specific information. Use the node navigation menu to select and configure a node.

Group	Associated implementations	Parameter	Description
Global resources	All	T.38	Indicates whether T.38 fax functionality is enabled.
	All	SIP info	Indicates whether the Programmable Media Platform can accept incoming SIP INFO messages with DTMF content.

Group	Associated implementations	Parameter	Description
	All	SIP to SIP server	Indicates whether the Programmable Media Platform can make SIP to SIP calls in passthrough mode; that is, with no audio transcoding.
	Implementations with licensed video transcoding ports	Video transcoding	Indicates whether video transcoder resources are available for the Programmable Media Platform. If enabled, video transcoding is inserted in the video path if the Programmable Media Platform finds incompatible video codec characteristics between call legs. If disabled, the call is dropped if the Programmable Media Platform finds incompatible video codec characteristics.
	Implementations with licensed video transcoding ports	Use video transcoding in dialog connections	Forces video transcoding in the dialog leg. If enabled, video transcoding is inserted in the video path regardless of the negotiated video codec on either side. If disabled, video transcoding only applies to calls with incompatible video codec characteristics.
Video Transcoder	Implementations with licensed video transcoding ports	Add video transcoder	Specifies the IP address of the video transcoder system to be used by the Programmable Media Platform. Click **Add video transcoder** to add this video transcoder system to the configuration. The IP address is added to the table as well as the system name and number of channels.
		ID	ID of the video transcoder system.
		IP address	IP address of the video transcoder system.

Group	Associated implementations	Parameter	Description
		Name	Video transcoder system name.
		Channels	Number of full-duplex video transcoder channels available for use by the Programmable Media Platform.
		Edit	Click **Remove** to remove the video transcoder system associated with this IP address from the configuration.
Resource	All	G.711	Indicates whether G.711 mu-law or G.711 A-law is enabled.
	Implementations with a G.723 license from Dialogic	G.723	Indicates whether G.723 is enabled.
	All	G.726-32	Indicates whether G.726 is enabled.
	Implementations with a G.729 license from Dialogic	G.729	Indicates whether G.729 is enabled.
	Video models, or audio models with an AMR license from Dialogic	AMR	Indicates whether AMR is enabled.
	Video models	Clear channel	Indicates whether clear channel is enabled.
	Video models	Mobile video	Indicates whether 3G-324M is enabled.
	Implementations with licensed conferencing ports	Conferencing	Indicates whether conferencing functionality is enabled.

Note: Do not use Advanced mode without contacting Dialogic Technical Services and Support.

SIP parameters

The SIP parameters page configures SIP-related settings including SIP load balancing. You can access this page by clicking **SIP** on the Configuration menu.

For a Vision node, this page displays server-specific information. Use the node navigation menu to select and configure a node.

Group	Associated implementations	Parameter	Description
General	All	Transport protocol	Defines whether the VoiceXML interpreter defaults to using TCP or UDP.
	All	SIP load balancing	Click the Enabled check box to enable SIP load balancing.
SIP load balancing servers	Implementations with SIP load balancing	Name	Name of server to be used as a SIP destination.
		IP address	IP address of the server to be used as a SIP destination.
		Port	Port for this server.
		Edit	Click **Remove** to remove this server.
		Add	Click **Add** to add a server.
Authentication	All	Realm	Associates a user name and password pair to a certain context. Used for SIP realm authentication. For example: dialogic.com.
		User name	User name associated with a realm.
		Password	Password for this user name.
		Edit	Click **Remove** to remove a user name and password pair.

Group	Associated implementations	Parameter	Description
		Add	Click **Add** to add a user name and password pair for a specific realm.
		Add Default	If the user name and password pair is to be used for all authentication requests regardless of the realm, click **Add Default**.

RTP parameters

The RTP parameters page configures media board settings. You can access this page by clicking **RTP** on the Configuration menu.

For a Vision node, this page displays server-specific information. Use the node navigation menu to select and configure a node.

Group	Associated implementations	Parameter	Description
Board #0 interface configuration	All	Interface	Ethernet interface, such as eth0 and eth1, or interface alias, such as eth0:1 and eth1:1, on the media board.
		Status	Status of the interface on the media board: enabled or redundant.
		IP address	IP address of the media board.
		Subnet mask	Subnet mask for the media board.
		Gateway	IP address of the router for the media board.
		VLAN	Virtual LAN (VLAN) ID, used to enable VLAN tagging for the media board.

Group	Associated implementations	Parameter	Description
		Edit	To create an alias for the selected interface, click **Add alias**. A new entry is added at the bottom of the table. To remove an interface definition, click **Remove**.
Board #1 interface configuration	Implementations where the Programmable Media Platform has two media boards		See Board #0 interface configuration for parameters and parameter descriptions.
Boards traffic types	All	Type	Type of traffic being sent through the media board: RTP or NbUP.
		Interface	Interface associated with the traffic type: RTP or NbUP.
Board #0 IP routes	All	IP address	IP address for the IP route of a configured interface.
		Subnet mask	Subnet mask for the IP route of a configured interface.
		Destination	Destination IP address for the IP route of a configured interface.
		Backup destination	Backup destination IP address for the IP route of a configured interface. Used when the network monitor service performs a failover or a switchover to the backup network.
		Edit	To remove an IP route definition, click **Remove**.
		Add	To create an IP route for a configured interface, click **Add**.

Group	Associated implementations	Parameter	Description
Board #1 IP routes	Implementations where the Programmable Media Platform has two media boards		See Board #0 IP routes for parameters and parameter descriptions. If this route is a clone of Board #0 IP routes, click the **Clone** field.

NbUP circuits

The NbUP page contains NbUP configuration information. Access this page by clicking **NbUP circuits** on the Configuration menu.

For a Vision node, this page displays server-specific information. Use the node navigation menu to select and configure a node.

Group	Parameter	Description
NbUP configuration	Mode	NbUP mode. Values are: SLAVE, MASTER, or MASTER if call originator.
	Frame duration	Frame size. Values are: 5 ms or 20 ms.
	PDU type	PDU type. Values are: CRC_ENABLE, CRC_DISABLE
BICC configuration	Group size	BICC circuit group size. Used when sending group circuit messages. This value should match the Circuit Group size used when defining circuit groups in the Signaling Server page.
Board	Board #	Media board number.
	Number of circuits	Number of circuits supported on the media board.
	Circuits	Read-only parameter. Range of circuits.
PSTN routes	Route #	Defines the route circuits reserved for outgoing PSTN calls.
	Strategy	Specifies how the circuits are selected for this route. For a description of the valid values, see Values for the PSTN routes group.

Group	Parameter	Description
	Circuit list	Range of circuits available for this route. Use comma-separated list of circuits or circuit range. For example: 1-128, 257-384, 387, 390
	Edit	Click **Remove** to remove a PSTN route.
	Add	Click **Add** to reserve another route circuit for outgoing PSTN calls.

Trunks

The Trunks page configures trunk settings and circuit groups. Access this page by clicking **Trunks** on the Configuration menu.

For a Vision node, this page displays server-specific information. Use the node navigation menu to select and configure a node.

Group	Associated Implementations	Parameter	Description
Trunk configuration	All	Frame type	Indicates frame type of T1 or E1.
		CRC signal checking	Indicates whether the media resource provides CRC signal checking.
		Line impedance	Type of cable connecting the media resource to the T1 or E1 network.
		Trunk framing	Framing format.
		Line code	Ones density maintenance method used on the trunk line to maintain a clear channel transmission.
		Clock source	Media board's clock source. • Network: source is from the PSTN. • Internal: source is from the internal oscillator.

Group	Associated Implementations	Parameter	Description
PSTN routes	All	Route 1 - Route ***n***	Defines the route circuits reserved for outgoing PSTN calls, and specifies how the circuits are selected. For a description of the valid values, see Values for the PSTN routes group. Click **Add** to reserve another route circuit for outgoing PSTN calls.
Board #0 trunks	All	Trunk 1 - Trunk***n***	Whether the specified trunk on Board 0 is enabled or disabled, and the PSTN route associated with the trunk.
Board #1 trunks	Implementations where the Programmable Media Platform has two media boards	Trunk 1 - Trunk***n***	Whether the specified trunk on Board 1 is enabled or disabled, and the PSTN route associated with the trunk.

Values for the PSTN routes group

Use a circuit code identifier (CCI) to identify a PSTN route. A CCI is an integer ranging from 1 to the highest circuit (CCI max). For example, if the routes contain four E1 trunks (each containing 30 circuits), circuits are numbered from 1 to 120. The value of CCI max is 120.

The following table describes the valid values for each route in the PSTN routes group:

Value	Description
FROM_TOP	Selects the first idle circuit in decreasing CCI order. This strategy always selects the highest available circuit. **Example** With this circuit selection strategy, a series of calls might be placed as follows: 1. A first call is placed on the last circuit, CCI max. 2. A second call is placed on circuit (CCI max – 1), because CCI max is busy processing the first call. 3. The first call terminates, so CCI max becomes idle. 4. A third call is placed on CCI max, because CCI max is now available. 5. A fourth call is placed on (CCI max - 2), because both CCI max and (CCI max - 1) are busy processing calls 3 and 2, respectively.
FROM_BOTTOM	Selects the first idle circuit in increasing CCI order. This strategy always selects the lowest available circuit. **Example** With this circuit selection strategy, a series of calls might be placed as follows: 1. A first call is placed on the first circuit, CCI 1. 2. A second call is placed on the second circuit, CCI 2, because the first circuit is busy processing the first call. 3. The first call terminates, so the first circuit becomes idle. 4. A third call is placed on CCI 1, because CCI 1 is now available. 5. A fourth call is placed on CCI 3, because CCI 1 and CCI 2 are busy processing calls 3 and 2, respectively.

Value	Description
DESCENDING	Selects a circuit by rotating circuits in decreasing CCI order, from the highest circuit (CCI max) down to the middle of the route ((CCI max / 2) + 1). If no circuit is idle on the second half of the route, a circuit on the first half of the route is selected. **Example** With this circuit selection strategy, a series of calls might be placed as follows: 1. A first call is placed on the last circuit, CCI max. 2. A second call is placed on (CCI max – 1). 3. The first call terminates, so CCI max becomes idle. 4. A third call is placed on (CCI max – 2). 5. For each subsequent call, the next lower circuit is selected up to the middle of the route. When the last circuit in the half route is reached ((CCI max / 2) + 1), the selection strategy rotates back to the last circuit CCI max, because that is the first available circuit in decreasing order of CCI.
ASCENDING	Selects a circuit by rotating circuits in increasing CCI order, from the first circuit (CCI 1) up to the middle of the route (CCI max / 2). If no circuit is idle on the first half of the route, a circuit on the second half is selected. **Example** With this circuit selection strategy, a series of calls might be placed as follows: 1. A first call is placed on the first circuit, CCI 1. 2. A second call is placed on the second circuit, CCI 2. 3. The first call terminates, so the first circuit becomes idle. 4. A third call is placed on CCI 3. 5. For each subsequent call, the next higher circuit is selected, up to the middle of the route. When the last circuit in the half route is reached (CCI max / 2), the selection strategy rotates back to CCI 1, because that is the first available circuit in increasing order of CCI.
TIMER	(Default) The selected circuit is the one on which the inactivity timer is the most important. At the beginning, all circuits have the same inactivity timer. The circuits are selected in decreasing CCI order, starting from CCI max down to 1. When all circuits have been used once, they are selected by the inactivity timer.

PSTN

The PSTN page contains additional configuration settings for the following Programmable Media Platform models:

- ISDN models
- ISUP models

Access this page by clicking **PSTN** on the Configuration menu.

For a Vision node, this page displays server-specific information. Use the node navigation menu to select and configure a node.

Additional settings for ISDN models

The following settings apply to the ISDN audio and ISDN video models:

Group	Parameter	Description
ISDN	ISDN type	ISDN protocol variant.
	ISDN equipment	ISDN equipment type.

For more information, see ISDN models.

Additional settings for ISUP models

The following settings apply to the ISUP audio and ISUP video models:

Group	Parameter	Description
Signaling servers	ID	Signaling server ID.
	Redundant pair	If checked, the signaling server is part of a redundant pair.
	IP	IP address for the signaling server or the signaling server pair.
	Circuit start	Starting number of the circuit that the signaling server or signaling server pair can handle.
	Switch type	ISUP protocol variant for the signaling server or signaling server pair, such as ETSIV2, ETSIV3, ANSIBICC, and ITUBICC.
	Point code	Point code for the signaling server or signaling server pair. Specify the point code value as a decimal or hexadecimal number.
	Edit	Click **Remove** to remove the associated signaling server from the list.

Trunk - Signaling Server association	Trunk	T1 or E1 trunk handled by the previously defined signaling servers. The values in this field vary, depending on the number of boards and the trunk configuration.
	Signaling Server	Signaling server ID.
Circuit - Signaling Server association (for BICC switch type only)	BICC circuit	BICC circuit configured on a Dialogic® CG Series Media Board. BICC circuits are grouped by media boards. The values in this field vary, depending on the number of media boards configured for BICC.
	Signaling Server	Signaling server ID.

For more information, see ISUP models.

Signaling Server

The Signaling Server page contains additional configuration information for ISUP models that have a Signaling Server. The information on this page varies with the configuration, such as redundant pair configuration, and MTP or SIGTRAN transport protocol. Access this page by clicking **Signaling Server** on the Configuration menu.

Group	Associated implementations	Parameter	Description
Signaling Server redundant pair configuration	All ISUP	Redundant pair	If checked, the signaling server is part of a redundant pair.
		Mate signaling IP	IP address for the mate signaling server.
		Mate OA&M IP	IP address for the mate signaling server OA&M interface.
		Server Role	SS701 is the default primary server. SS702 is the default backup server.
TX Board Redundant IP information	Implementations with redundant servers	IP address	IP address of the Dialogic® TX 5000 Series SS7 Board network interface used for redundancy.
		Subnet mask	Subnet mask of the TX 5000 Series SS7 Board network interface used for redundancy.

Group	Associated implementations	Parameter	Description
		Mate IP address	IP address of the other TX 5000 Series SS7 Board used in the redundancy pair.
ISUP & MTP base configuration	All ISUP	Node point code	Point code for the signaling server or signaling server pair. Specify the point code value as a decimal or hexadecimal number.
		SS7 subservice field	MTP 3 subservice.
ISUP base configuration	All ISUP	ISUP switch type	ISUP protocol variant for the Signaling Server or Signaling Server pair, such as ETSIV2, ETSIV3, ANSIBICC, and ITUBICC.
MTP base configuration	Implementations where transport is MTP	Transport	Indicates transport protocol: MTP or SIGTRAN.
		MTP link type	MTP 3 protocol variant.
		MTP links	Indicates how MTP links for ISUP arrive at the gateway. When checked, the links are embedded in the trunks connected to the media boards. When unchecked, the links are connected directly to the signaling board.
M3UA base configuration	Implementations where transport is SIGTRAN	Transport	Indicates transport protocol: MTP or SIGTRAN.
		Local routing context	Used when transport is set to SIGTRAN.
		Network appearance code	Values are determined and configured by network operators on each side of an association.
		Service variant	Protocol variant of the M3UA service user.
		Source port	Listening STCP port.

Group	Associated implementations	Parameter	Description
		DPC length	Destination point code length.
TX Board SIGTRAN IP information	Implementations where transport is SIGTRAN	IP address	IP address of the Dialogic® TX 5000 Series SS7 Board.
		Subnet mask	Subnet mask of the TX 5000 Series SS7 Board.
		Gateway	IP address of the router for the TX 5000 Series SS7 Board.
Peer signaling process	Implementations where transport is SIGTRAN	ID	ID of the peer signaling process.
		IP	IP address of the peer signaling process.
		Port	Remote SCTP port.
		PS type	Peer signaling type: IPSP or SGP.
		IPSP mode	Valid when PS type is IPSP. Indicates whether the IPSP mode is single-ended or double-ended.
		Dynamic routing key management	Indicates whether this peer signaling process can send and receive dynamic routing key management (DRKM) messages.
		Use network appearance	Determines whether the optional network appearance parameter is included when communicating with the remote peer.
		ASP	Indicates whether an ASP identifier is required in sent and/or received ASPUP and ASPUP ACK (ASP Up Acknowledgement) messages.

Group	Associated implementations	Parameter	Description
		Client	For PS type of IPSP, checked indicates that associations are automatically initiated from this PSP. For PS type of IPSP, unchecked means associations are not initiated from this PSP. The other side is expected to initiate any associations.
		Edit	Removes the peer signaling process.
Destination point codes	All ISUP	DPC	Destination point code for a circuit.
		Adjacent	Specifies whether the point code is adjacent to the Programmable Media Platform.
		Routing context	For SIGTRAN transport type, specifies the routing context of the peer server.
		Associated PSP	For SIGTRAN transport type, specifies the space-separated peer signaling process(es) associated with the DPC.
		Edit	Removes the associated destination point code from the list.
		Add new DPC	Enter the destination point code for another circuit, and click **Add New DPC** to add another destination point code to the list.
ISUP circuits definition	All ISUP	Index	Circuit group index.
		Start circuit	Starting number of the circuit group that the signaling server or signaling server pair can handle.

Group	Associated implementations	Parameter	Description
		Start CIC	Starting CIC of the circuit group that the signaling server or signaling server pair can handle.
		Number of circuits	Number of circuits in the circuit group.
		DPC	Destination point code associated with the circuit group.
		Unused circuits	A space-separated list of circuits within the range of this circuit group that are not controlled by ISUP.
		Edit	Removes the current circuit definition.
MTP links	Implementations where transport is MTP	Trunk	T1 or E1 trunk used for signaling. The values in this field vary, depending on the number of boards and the trunk configuration.
		Status	If checked, indicates that the MTP link is enabled.
		Adjacent DPC	Destination point code associated with each route.
		Other DPC	A space-separated list of non-adjacent DPCs reachable by this link.
		SLC	Signaling link code.
		Signaling timeslot	Timeslot on the signaling trunks to be reserved for signaling. For E1 line types, the value is usually 16.
		Speed	Speed of signaling link in Kbps.
Remote MTP	Implementations	Index	Index of the link definition.

Group	Associated implementations	Parameter	Description
links	with redundant servers	Port Number	Remote TX 5000 Series SS7 Board port number (corresponds to the MTP link index configured on the remote server).
		DPC	Destination point code of the remote link.
		Other DPC	A space-separated list of non-adjacent DPCs reachable by this link.
		SLC	Signaling link code of the remote link.
		Edit	Removes the current remote link definition.

For more information, see ISUP models.

Options

The Options page contains global and advanced settings. Access this page by clicking **Options** on the Configuration menu.

Group	Parameter	Description
Global	Billing	If checked, billing is enabled.
Advanced Telecom	Parameters in drop-down list	Select a parameter to be configured from the drop-down list and click **Add**. The parameter is displayed in the window above and its value can be edited. For more information on these parameters, see the *Dialogic® Vision™ Call Server Administration Manual*.

Capacity upgrade

The Capacity upgrade page lets you configure the port capacity of the Programmable Media Platform. Access this page by clicking **Capacity** on the Configuration menu.

For a Vision node, this page displays server-specific information. Use the node navigation menu to select and configure a node.

Parameter	Description
New VoiceXML Interpreter port rating	Port capacity of the VoiceXML Interpreter based on the number of licenses purchased.

Parameter	Description
Announcement port rating	Port capacity for announcements; can be used for Video Call Completion to Voice (VCCV) feature and playback of network announcements feature.
Call Server	Port capacity of the Call Server based on the number of licenses purchased.

SNMP configuration

The SNMP configuration page lets you configure SNMP parameters for the Programmable Media Platform. Access this page by clicking **SNMP** on the Configuration menu.

Group	Parameter	Description
SNMP Base Configuration	Version	SNMP version.
	Engine ID	For SNMPv3, Engine ID.
System Description	Name	System name.
	Description	System description.
	Location	System location.
	Contact	Whom to call when the system needs attention.
Communities (SNMPv1 and SNMPv2c only)	Read only community	Read-only community name.
	Read/Write community	Read-write community name.
Users (SNMPv3 only)	Name	User name. Add user name in the field and click **Add new user**. The user name is shown in the Name field.
	Permission	Permission type: read-only or read/write.
	Authentication Password	Password and password type (MD5 or SHA).
	Privacy Password	Password and password type (DES or AES).
	Edit	To remove a user, click **Remove**.
Traps Receivers	Receiver IP	IP address for trap receiver.

	Port	Port for receiver IP.
	Community	For SNMPv1 and SNMPv2c, default trap sink community to use.
	User	For SNMPv3, user name.
	Edit	To remove a trap receiver, click **Remove**.
	Add	To add a trap receiver, click **Add**.

Network redundancy configuration

The Network Redundancy Configuration page lets you configure network redundancy parameters. Access this page by clicking **Network redundancy** on the Configuration menu.

For a Vision node, this page displays server-specific information. Use the node navigation menu to select and configure a node.

Group	Parameter	Description
Redundancy Manager	Network redundancy	Click the check box to enable network redundancy. Leave blank to disable.
	Monitoring frequency	Monitoring frequency in milliseconds.
	Monitoring timeout	Number of times the network monitor service pings the monitored interfaces before it triggers a failover.
Monitored Interfaces	Location	Location of the monitored interface, such as Host or Board 0.
	Name	Name of the monitored interface, such as bond0 or eth0.
	Monitor	Click the check box to enable monitoring of this interface.
Monitored networks and IPs	Network name	Name of the monitored network, such as Signaling or Billing.
	Primary IP(s)	IP address or addresses to monitor on the primary network. Separate multiple IP addresses with a space.

Group	Parameter	Description
	Backup IP(s)	IP address or addresses to monitor on the backup network. Separate multiple IP addresses with a space. If blank, the network monitor service uses the same address for the primary network and the backup network.
	Originator	Origin of the network monitoring. Values are: All, Media board(s), and Host. For example, if set to Media boards, the network will be monitored from the boards. If set to Host, the network will be monitored from the host. If set to All, the network will be monitored from everywhere.
	Edit	Click **Remove** to remove this network and associated IP addresses from being monitored.
	Add	Click **Add** to add a network and associated IP addresses to be monitored.
Virtual IP addresses	Interface	Interface associated with the virtual IP address to be used for single SIP entry point in SIP load balancing. This interface is defined on the Host IP information page.
	Peer IP	Peer IP address with which the virtual IP address is shared.
	Monitored Network	Name of a previously defined monitored network.
	Edit	Click **Remove** to remove this virtual IP address from being used in SIP load balancing.
	Add	Click **Add** to add a virtual IP address.

For more information about network redundancy, see Network redundancy and the network monitor service. For more information about SIP load balancing, see Using SIP load balancing.

Video Transcoder

The Video Transcoder page lets you configure video transcoder resources for a video transcoder system. Access this page by clicking **Video Transcoder** on the Configuration menu.

Parameter	Description
Channels	Number of full-duplex video transcoder channels that are available for this system.
Usage high water (%)	High water mark for CPU usage in percentage. If this threshold is reached, the system issues an SNMP notification.
Usage low water (%)	Low water mark for CPU usage in percentage. If this threshold is reached, the system issues an SNMP notification to indicate that the CPU level has returned to an acceptable level. This notification only occurs if the high water mark notification was previously issued.
Usage reject high water (%)	Upper limit of high water mark for CPU usage in percentage. If this threshold is reached, the system issues an SNMP notification and begins to reject calls.
Usage reject low water (%)	Lower limit of high water mark for CPU usage in percentage. If this threshold is reached, the system issues an SNMP notification and begins to accept calls.

For more information about video transcoding, see Managing video transcoder resources.

Import/Export configuration

The Import/Export configuration page lets you back up and restore a Programmable Media Platform configuration. Access this page by clicking **Import/Export** on the Configuration menu. For more information, see Backing up a configuration and Restoring a configuration.

Operations menu parameters

The Operations menu contains the following pages:

- Services
- Maintenance

Services

Use the Services page to start or stop Programmable Media Platform services, quiesce the Call Server, VoiceXML Interpreter or Video Transcoder, and set up services to start up automatically when the Programmable Media Platform starts up. Access this page by clicking **Services** on the Operations menu.

Basic Services page

The basic Services page contains the following parameters.

Field	Description
Status	Indicates the server status.

Field	Description
Managed services	Indicates the number of managed services.
Started services	Indicates the number of services that have been started.

Advanced Services page

The advanced Services page provides more detail on each service. This page contains the following parameters. Each row in the table represents one service.

Field	Description
Service name	Name of the service.
Status	Indicates whether the service is starting, started, stopped, quiesced, or unavailable. Unavailable is for node members only. Quiesced is for Call Server, VoiceXML Interpreter, and Video Transcoder only. The Programmable Media Platform can only accept new calls when the service is started.
Managed	Specifies how the service starts. When selected, the service is started automatically on start-up. If not selected, you must start the service manually.
Tasks	One of the following actions: • Start: Starts the service. • Stop: Stops the service. • Restart: Stops and then restarts the service. • Quiesce: Stops the service from processing new calls. Quiesce is for Call Server, VoiceXML Interpreter, and Video Transcoder only. Quiesce does not affect calls that are currently being processed.

Maintenance

Use the Maintenance page to change the log levels of Programmable Media Platform logs, clear the application data cache, deploy licenses, and install patches. Access this page by clicking **Maintenance** on the Operations menu.

For a Vision node, each node member and associated information is displayed in its own server-specific page. Actions on a page affect the selected node member only.

Group	Field	Description
Logging settings	Service name	Name of the service.

Group	Field	Description
	Log level	Log level of the associated service. Valid values are as follows, in order of decreasing severity and increasing verbosity: • FATAL • ERROR • WARNING • INFO1 • INFO2 • INFO3 • INFO4 • INFO5
	Cache	Click **Cache** to clear the application data cache for the associated service.
Component states	Component	Name of the component, such as Signaling Server or Ethernet Redundancy.
	Status	Indicates the status of the component. For example, indicates whether the Signaling Server being managed is a standalone server, the primary server or backup server.
	Action	Click **Switch** to change the current server role.
N/A	Software RAID	If RAID status is degraded, you can click **Rebuild** to rebuild a RAID-1 array disk.
N/A	Deploy license	For information about deploying licenses, contact Dialogic Technical Services and Support.
N/A	Install patch	For information about installing patches, contact Dialogic Technical Services and Support.

Provisioning menu parameters

The Provisioning menu contains the following pages:

- Routing profiles configuration
- Call routing table
- CCXML application configuration
- VoiceXML application configuration
- Video transcoder resource configuration

Routing profiles configuration

The Routing profiles configuration page specifies the profile of a route. Access this page by clicking **Gateway profiles** on the Provisioning menu. For information about defining a routing profile, see the *Dialogic® Vision™ 1000 Video Gateway Administration Manual*.

Call routing table

The Call Routing Table page defines routes for the Programmable Media Platform using the gateway routing table. This table is used when the Call Server is configured to work as a gateway. Access this page by clicking **Gateway routes** on the Provisioning menu. For more information, see the *Dialogic® Vision™ 1000 Video Gateway Administration Manual*.

CCXML application configuration

The CCXML application configuration page defines custom CCXML applications to the Programmable Media Platform. Access this page by clicking **CCXML applications** on the Provisioning menu. For more information, see Managing CCXML applications.

VoiceXML application configuration

The VoiceXML application configuration page defines VoiceXML applications to the Programmable Media Platform. Access this page by clicking **VoiceXML applications** on the Provisioning menu. For more information, see Managing VoiceXML applications.

Video transcoder resource configuration

The Video transcoder resource configuration page defines video transcoder resources for the Programmable Media Platform. Access this page by clicking **Video transcoder resources** on the Provisioning menu.

To enable video transcoding, see the Resource configuration page on the Configuration menu. For more information about video transcoding, see Managing video transcoder resources.

Parameter	Description
ID	ID of the video transcoder system.
IP address	IP address of the video transcoder system to be used by the Programmable Media Platform. Enter the IP address and click **Add video transcoder** to add this video transcoder system to the configuration. The IP address is added to the table as well as the system name and number of channels.
Name	Video transcoder system name.
Channels	Number of full-duplex video transcoder channels available for use by the Programmable Media Platform.
Edit	Click **Remove** to remove the video transcoder system associated with this IP address from the configuration.

Monitoring menu parameters

The Monitoring menu contains the following pages:

- RAID page
- Trunks page
- CCXML statistics
- Call Server status
- VoiceXML view
- VoiceXML status
- SSML Processor view
- SSML Processor status
- Signaling Monitor
- Video Transcoder status
- Network Monitor
- Log files
- CDR files

Note: If your environment includes multiple Vision Servers defined as a Vision node, some pages contain information that is common to all node members, while other pages contain server-specific information. Use the node navigation menu to select a node member.

RAID page

The RAID page displays RAID status. Access this page by clicking **RAID** on the Monitoring menu.

Group	Field	Description
RAID-1 Array	Volume status	Volume status: optimal or degraded. Optimal indicates both disk drives are online. Degraded indicates one or both disk drives are missing.
	Rebuild percentage	Indicates progress of RAID-1 array rebuild in percentage.
Disk 0	Status	Status of the first disk: online, missing, or offline requested.
	Additional flag	Additional information if any. For example, shows out-of-sync if rebuild is in progress.
Disk 1	Status	Status of the second disk: online, missing, or offline requested.
	Additional flag	Additional information if any. For example, shows out-of-sync if rebuild is in progress.

If you replace a disk drive, the Vision Console shows the new drive's status as offline requested. To rebuild the new drive, go to the Maintenance page on the Operations menu and click **Rebuild** next to RAID. The RAID page is then displayed showing the new drive as online and out-of-sync, and the rebuild percentage in progress.

Trunks page

The Trunks page displays a trunk monitoring chart. Access this page by clicking **Trunks** on the Monitoring menu. If your Dialogic® Vision™ Server is configured to use BICC, this page is called **Circuits** on the Monitoring menu.

Field	Description
Circuits	Total number of configured circuits.
Circuit states	Information about circuit states. • Idle: Circuit not used • Busy • Out of service • Blocked • Unknown
Trunk states	(Not available for BICC) Information about trunk (line) states. • In sync • Yellow alarm • Blue alarm • Red alarm
Signaling	(ISUP models) Number of circuits reserved for signaling (raw circuits). The Vision Console displays this information as you hover over an unknown circuit in the chart.
Blocked	Information about blocked circuits, if any. The Vision Console displays this information as you hover over a blocked circuit in the chart. Valid values: • local-maintenance: Circuit is locally blocked for maintenance. • remote-maintenance: Circuit is remotely blocked for maintenance. • local-hardware: Hardware is locally blocked. • remote-hardware: Hardware is remotely blocked.

CCXML statistics

The CCXML statistics page displays a bar chart that represents the following statistics:

- Number of active CCXML sessions, connections, and dialogs.
- Maximum number of CCXML sessions, connections, and dialogs.

Access this page by clicking **CCXML statistics** on the Monitoring menu.

Call Server status

The Call Server status page displays status information for the Call Server. Access this page by clicking **Call Server status** on the Monitoring menu.

<table>
<tr><th>Group</th><th>Description</th></tr>
<tr><td>Server information</td><td>Contains the following fields:
<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>Status</td><td>Status of the Call Server:
Loading Telecom: server is starting.
Online: server started and is accepting incoming calls.
Quiesced: server is quiesced and rejects incoming calls.</td></tr>
<tr><td>Active SIP calls</td><td>Number of active SIP calls.</td></tr>
<tr><td>Active PSTN calls</td><td>Number of active PSTN calls.</td></tr>
<tr><td>Total calls</td><td>Total number of calls since process startup.</td></tr>
<tr><td>Max concurrent calls</td><td>Maximum number of concurrent calls since process startup.</td></tr>
</table></td></tr>
</table>

<table>
<tr><th>Group</th><th>Description</th></tr>
<tr><td>System information</td><td>Contains the following fields:
<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>Version</td><td>Name and version of the Call Server, with the process name in brackets.</td></tr>
<tr><td>Startup time</td><td>Time the process started in coordinated universal time (UTC).</td></tr>
<tr><td>Uptime</td><td>Total time in days, hours, and minutes since process startup.</td></tr>
<tr><td>Channel</td><td>Configured number of channels accepting calls, available to place calls, or both.</td></tr>
<tr><td>Licenses</td><td>Available number of Call Server licenses.</td></tr>
</table></td></tr>
<tr><td>CCXML statistics</td><td>Displays the following statistics:
Number of active CCXML sessions, connections, and dialogs.
Maximum number of CCXML sessions, connections, and dialogs.</td></tr>
<tr><td>Cache information</td><td>Displays the current and maximum memory usage and disk usage of the Call Server's internal caches. The server has the following types of caches:
CCXML, for CCXML scripts.
Script, for JavaScript files fetched from a CCXML script.
When the limit is reached on a cache, older and less frequently used resources are deleted from the cache.</td></tr>
<tr><td>Channel information</td><td>Displays the current execution (or health) status for each channel belonging to the server.</td></tr>
</table>

VoiceXML Interpreter view

The VoiceXML Interpreter view page displays a real-time view of the VoiceXML Interpreter's current activity. Access this page by clicking **VoiceXML view** on the Monitoring menu.

Field	Description
[online] [offline]	Status of the VoiceXML Interpreter: • [online] if the channels are accepting new connections. • [offline] if the channels have been quiesced. For information on quiescing and restarting channels, see Working with Programmable Media Platform services.
calls in progress	Number of calls currently in progress.
calls total	Total number of calls since startup.
max concurrent calls	Maximum number of concurrent calls since startup.

The VoiceXML Interpreter view page also lists the channels as they are used, including the DNIS and ANI of each call.

VoiceXML Interpreter status

The VoiceXML Interpreter status page displays VoiceXML Interpreter status information. Access this page by clicking **VoiceXML status** on the Monitoring menu.

Group	Description
Session information	Contains the following fields:

Field	Description
Status	Status of the VoiceXML Interpreter: • [online] The channels are accepting new connections. • [offline] The channels are quiesced.
Calls in progress	Current number of calls.
Total calls	Total number of calls since startup.
Max concurrent calls	Maximum number of concurrent calls since process startup.

<table>
<tr><th>Group</th><th>Description</th></tr>
<tr><td>System information</td><td>Contains the following fields:
<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>Version</td><td>Name and version of the VoiceXML Interpreter, with the process name in brackets.</td></tr>
<tr><td>Configuration file</td><td>Name of the configuration file for the current process.</td></tr>
<tr><td>MIO</td><td>Name and version of the current Media Interface Object implementation. The Media Interface Object is used by the Media Resource Function component of the VoiceXML subsystem.</td></tr>
<tr><td>Startup time</td><td>Time the process started in coordinated universal time (UTC).</td></tr>
<tr><td>Uptime</td><td>Total time in days, hours, and minutes since process startup.</td></tr>
<tr><td>Channels</td><td>Configured number of channels accepting calls, available to place calls, or both.</td></tr>
<tr><td>Licenses</td><td>Available number of licenses for the different call modes.</td></tr>
</table></td></tr>
<tr><td>Health information</td><td>Displays information about the health of the channels and identifies any channel that has generated an exception or is not responding. OK displays if all channels are operating normally.</td></tr>
<tr><td>Cache information</td><td>Displays the current and maximum allowed sizes of the VoiceXML Interpreter's internal caches. The VoiceXML Interpreter has caches for:
VoiceXML documents
Grammar files
ECMA script files
XML documents
When the limit is reached on a cache, older and less frequently used resources become eligible for deletion from the cache.</td></tr>
</table>

Group	Description
Channel information	Displays the current execution (or health) status for each channel belonging to the VoiceXML Interpreter.

SSML Processor view

The SSML Processor view page displays a real-time view of the SSML Processor's current activity. Access this page by clicking **SSML view** on the Monitoring menu.

Field	Description
[online] [offline]	Status of the SSML Processor: • [online] The channels are accepting new connections. • [offline] The channels are quiesced. For information about quiescing and restarting channels, see Working with Programmable Media Platform services.
current client connections	Number of calls currently in progress.
max concurrent client connections	Maximum number of concurrent calls since start-up.

The SSML Processor view page also displays:

- Information about each client as it connects and disconnects, including the client IP address.
- Text-to-speech (TTS) information including the vendor name, locale name, server address, server port number, and channel number.

SSML Processor status

This SSML Processor status page displays SSML Processor status information. Access this page by clicking **SSML status** on the Monitoring menu.

<table>
<tr><th>Group</th><th>Description</th></tr>
<tr><td>Session information</td><td>Displays the following types of session information:
<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>Status</td><td>Status of the SSML Processor:
[online] The channels are accepting new connections.
[offline] The channels are quiesced.</td></tr>
<tr><td>Current client connections</td><td>Current number of connections.</td></tr>
<tr><td>Max concurrent client connections</td><td>Maximum number of concurrent client connections since process startup.</td></tr>
</table></td></tr>
<tr><td>System information</td><td>Displays the following types of system information:
<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>Version</td><td>Name and version of the SSML Processor, with the process name given in brackets.</td></tr>
<tr><td>Configuration file</td><td>Name of the configuration file for the current process.</td></tr>
<tr><td>Audio driver</td><td>Name and version of the current audio driver (AD) implementation.</td></tr>
<tr><td>Startup time</td><td>Time, in coordinated universal time (UTC), at which the process was started.</td></tr>
<tr><td>Uptime</td><td>Total time in days, hours, and minutes since process startup.</td></tr>
<tr><td>Channels</td><td>Total number of SSML Processor channels.</td></tr>
</table></td></tr>
<tr><td>Health information</td><td>Displays information about the health of the channels and identifies any channel that has generated an exception or is not responding. OK displays if all channels are operating normally.</td></tr>
</table>

Group	Description
Cache information	Displays the current and maximum allowed sizes of the prompt and lexicon caches. The memory cache size and the disk cache size are displayed.
Statistics information	Provides statistical information about TTS engine usage, giving the total usage, maximum concurrent usage, and number of available instances for each locale. An administrator uses this information to determine if the number of instances available for a particular locale is sufficient for the current load.

Signaling Monitor

The Signaling Monitor page displays status information for a Signaling Server if one is used. Access this page by clicking **Signaling Monitor** on the Monitoring menu.

Group	Field	Description
Signaling server status	Server ID	ID of the signaling server.
	IP	IP address for the signaling server.
	Role	Indicates whether the server acts in a primary or secondary role.
	Location	Indicates whether the server is local or remote.
	Status	Status of the signaling server.
Signaling links status	Link index	MTP link identifier.
	Server ID	ID of the signaling server owning this link.
	Status	MTP link status.

Video Transcoder status

The Video Transcoder status page displays status information for the video transcoder.

Group	Field	Description
Server information	Active channels	Number of full-duplex video transcoder channels currently in use.

Group	Field	Description
	Active gateways	Number of gateways currently connected to this video transcoder.
	Usage level	Current CPU usage level of the video transcoder system.
	Errors	Number of errors generated since the video transcoder was last started.
	Warnings	Number of warnings generated since the video transcoder was last started.
System information	Version	Version of the video transcoder.
	Startup time	Time the video transcoder was last started in coordinated universal time (UTC).
	Uptime	Total time in days, hours, and minutes since the video transcoder was last started.
	Channels	Maximum number of full-duplex video transcoder channels that have been configured for use on this system. **Note:** The number of channels configured for use may be less than or equal to the maximum allowed by the license. The number of channels available for the system is defined in the Video Transcoder page of the Configuration menu.
Active gateway information	ID	ID of the gateway.
	State	Current overall state of the gateway.
	Name	Name of the application running on the gateway.
	Host	Host name of the gateway.
	Channels	Number of full-duplex channels in use for the active gateway.
	Startup time	Time the gateway was connected to the video transcoder in coordinated universal time (UTC).

Network Monitor

The Network Monitor page displays status information for the network including network redundancy. The information displayed on this page varies with the configuration. Access this page by clicking **Network** on the Monitoring menu.

Group	Field	Description
Network Interfaces	Name	Location and name of the monitored interface, such as Host eth0 or Board0 eth0.
	Status	Status of the monitored interface: UP or DOWN.
	IP	IP address of the monitored interface.
	Link Speed	Link speed of the monitored interface.
	Active Slave / Slave Role	Role of the monitored interface: Primary, Backup, or N/A.
General Redundancy State	Status	Redundancy status: Primary, Backup, or Deadlock. Primary means that all of the redundant pairs use the primary physical interface. Backup means that all of the redundant pairs use the backup physical interface. Deadlock means that the network monitor is unable to synchronize all of the interfaces; that is, the monitored IP addresses cannot be reached from either the primary or the backup interface.
Monitored Networks Status	Name	Name of the monitored network, such as Signaling or Billing.
	Status	Status of the monitored network: Available or Unavailable.

Log files

The Log files page displays log file information for the following components:

- Web Console
- Call server
- VoiceXML interpreter
- SSML processor
- Signaling server
- Video transcoder

- System service
- Process monitor
- Network monitor

To view a log file, click on the file name. To save the log file locally, click on **Save as**.

If you are logged in as root, the **Clear <component> logs** button at the bottom of the page allows you to delete all files for the current component.

CDR files

The CDR files page displays CDR file information for the following components:

- Call server
- VoiceXML interpreter

To view a CDR file, click on the file name. To save the CDR file locally, click on **Save as**.

If you are logged in as root, the **Clear <component> CDRs** button at the bottom of the page allows you to delete all files for the current component, except for the ones in the most recent directory. CDR files are accumulated in a directory structure where the year, month, and day is a different directory level. The directory for the most recent day is never deleted.

System menu parameters

The System menu contains the following pages:

- Authentication
- User administration

For more information, see User account management and Centralized user authentication.

Authentication page

The Authentication page contains information to configure the server's security environment. Access this page by clicking **Authentication** on the System menu.

Any action on this page automatically restarts the Vision Console and the local authentication server.

Group	Parameter	Description
Server certificate	Date created	Date that the server certificate was created.
	Expires	Date that the server certificate expires.
	Status	Status of the server certificate. Values are: Valid, Invalid.
	Create	This button is used to manually create a new self-signed certificate; for example, if you believe the certificate can no longer be trusted. Under normal conditions, you should not need to use this button.

Group	Parameter	Description
LDAP server setting	Type	Lightweight directory access protocol (LDAP) server setting. Values are: Provider, Consumer, Standalone.
	Read-only	Read-only setting for the Provider type. Values are: Yes, No.
N/A	Provider	For the Consumer type, specifies the IP address of the Provider server. Once set, the user will need to install the server certificate.
	Server status	Values are: Trusted, Untrusted, Unavailable.
	Action	Click **Install** to install the Provider server certificate on the Consumer server.

User administration page

The User administration page contains information about user accounts. Access this page by clicking **User administration** on the System menu.

Group	Parameter	Description
N/A	User name	Lists all user names in the system.
	Access level	Lists access level for each user name.
	Edit	Click **Edit** to edit user account information for a user name. Click **Reset Password** to reset the password to the system default.
Edit user	User name	Unique user name.
	Password	Password for this user name.
	Confirm password	Confirm password for this user name.
	Access level	Access level for this user name. Values are: root administrator, system administrator, application administrator, and guest.

5. Managing the Dialogic® Vision™ 1000 Programmable Media Platform

Working with Programmable Media Platform services

Use the Dialogic® Vision™ Console to set up auto-start for Programmable Media Platform services and to manually start, stop, and restart these services. The Vision Console also lets you quiesce the Call Server service, VoiceXML Interpreter service, and Video Transcoder service. This forces the service to stop processing new calls, but does not affect any calls currently being processed.

To work with Programmable Media Platform services:

Step	Action
1	Access the Vision Console, as described in Accessing the Dialogic Vision Console.
2	Click **Services** on the Operations menu. The Services page appears. Select **Advanced** from the drop-down list. On a Vision node, select the node member you want to manage. Actions on this page affect the selected node member only.
3	Click the **Managed** field to change the auto-start capability for the associated service.

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>4</td><td>Click one of the following buttons in the Tasks column for the entity you want to manually start, stop, quiesce, or restart:
<table>
<tr><th>Button</th><th>Description</th></tr>
<tr><td>Start</td><td>Starts the associated service. The service status changes to Starting while the service starts, and to Started after it is fully started.</td></tr>
<tr><td>Stop</td><td>Stops the associated service. The service status changes to Stopped when the service is stopped.
If you click Stop for the Call Server, VoiceXML Interpreter, or Video Transcoder service, the service stops without completing the in-process calls.</td></tr>
<tr><td>Quiesce</td><td>(Call Server, VoiceXML Interpreter, and Video Transcoder services only) Forces the service to stop processing new calls, but does not affect any calls currently being processed. Use this functionality to interrupt service and carry out maintenance operations on an active server without disturbing active calls.
Once maintenance operations are complete, you can stop, restart, or unquiesce the quiesced services. Unquiesce allows the server to accept new calls without restarting.</td></tr>
<tr><td>Restart</td><td>Stops and then restarts the associated service. If you restart the Call Server, VoiceXML Interpreter, or Video Transcoder service, the service stops without completing the in-process calls.
Click Restart All to stop and restart all server services.</td></tr>
</table></td></tr>
</table>

Once the Programmable Media Platform is in production, you should stop components only when you need to troubleshoot a call issue. You should stop components in the following order:

1. Call Server
2. Video Transcoder
3. VoiceXML Interpreter
4. SSML Processor

Always quiesce the Call Server, VoiceXML Interpreter, and Video Transcoder before you stop them, so that existing calls are not disrupted.

Viewing Programmable Media Platform information

Use the Dialogic® Vision™ Console to view information about Programmable Media Platform components and interfaces. The following table describes the procedures for viewing information in the Vision Console:

Component	Type of information	Procedure
Trunks (for PSTN models)	Illustration of trunk and circuit states, which shows: • The total number of circuits. • Whether a circuit is idle, busy, out of service, or blocked. • Whether a trunk is in sync or in an alarm state. If a trunk is in an alarm state, the Trunks page shows the alarm.	Click **Trunks** on the Monitoring menu or click **Circuits** if the server is configured for BICC. For field descriptions, see Trunks.
CCXML statistics	Number of active CCXML sessions, connections, connections, conferences, and dialogs.	Click **CCXML statistics** on the Monitoring menu. For field descriptions, see CCXML statistics.
Call Server	Server information, system information, CCXML statistics, cache information, and channel information.	Click **Call Server status** on the Monitoring menu. For field descriptions, see Call Server status.
VoiceXML Interpreter	Current activity	Click **VoiceXML view** on the Monitoring menu. For field descriptions, see VoiceXML Interpreter view.
	Status	Click **VoiceXML status** on the Monitoring menu. For field descriptions, see VoiceXML Interpreter status.
SSML Processor	Current activity	Click **SSML view** on the Monitoring menu. For field descriptions, see SSML Processor view.
	Status	Click **SSML status** on the Monitoring menu. For field descriptions, see SSML Processor status.

Audit tracking

The audit tracking tool allows the system administrator to identify user access and changes submitted through the Vision Console. The tool stores a copy of the system configuration

between various submittals and identifies the user who submitted the changes. The system administrator can use this information to determine whether to roll back to a previous known working configuration, for example, in case of a server failure.

The audit tracking tool monitors and logs the following types of activity on the Vision Console:

- User access
- Configuration changes from the Configuration menu
- Service update and maintenance updates from the Operations menu
- Provisioning changes from the Provisioning menu
- System-level changes from the System menu

Audit tracking console log files

Actions that change the status of the Video Gateway are logged in an audit tracking console log file. Examples of these actions as previously described are: user access, changes submitted in the Configuration, Operations, Provisioning, and System menu.

To view console log files, select the **Log files** option from the Monitoring menu and click the **Web Console** tab.

Console log files are available in HTML format and text file format. Text file formats may be requested by Dialogic Technical Services and Support for troubleshooting purposes. The naming convention for console log files is:

- *console0.html* to *console9.html*
- *console0.log* to *console9.log*

Up to 10 files of 10 MB each can be stored at one time. The file rolls over when the maximum size is reached; the oldest file is removed when the maximum limit is reached. These files are stored in */opt/nms/vx/cfgtool/webapps/WebConfigurator/logs/audit/*.

Audit tracking configuration archives

A configuration archive file is created when a user submits a change from the Configuration, Operations, Provisioning, or System menu. A separate configuration archive file is created for each of these activities.

To view configuration archive files, select the **Log files** option from the Monitoring menu and click the **Web Console** tab.

The configuration archive file stores the date, time, user name, and a configuration snapshot taken after the change was applied. Up to 500 configuration archive files per user can be stored at one time. The oldest archive is removed when the maximum limit is reached.

The system administrator has the option to download or restore a particular configuration directly from the archive file itself.

The naming convention for configuration archive files is:

YYYYMMDD-HHMMSS-user.zip

where *user* represents the user name who submitted the configuration change.

Configuration archive files are in ZIP format and are stored in */opt/nms/vx/cfgtool/webapps/WebConfigurator/bck/*.

Managing VoiceXML applications

Use the Dialogic® Vision™ Console to specify the following properties for individual VoiceXML applications:

- A number range that maps to the application. Incoming calls within the specified range are processed by the application.
- The initial URI to use for an incoming call, based on the specified number range.
- Whether logging is enabled, and the name of the application log file.
- Whether call transfers are allowed.
- The maximum call duration.

To specify properties for an application, first add the application to the Vision Console.

This topic describes how to:

- Add a VoiceXML application
- Remove a VoiceXML application
- Modify a VoiceXML application definition

It also describes the pattern matching syntax used in application definitions.

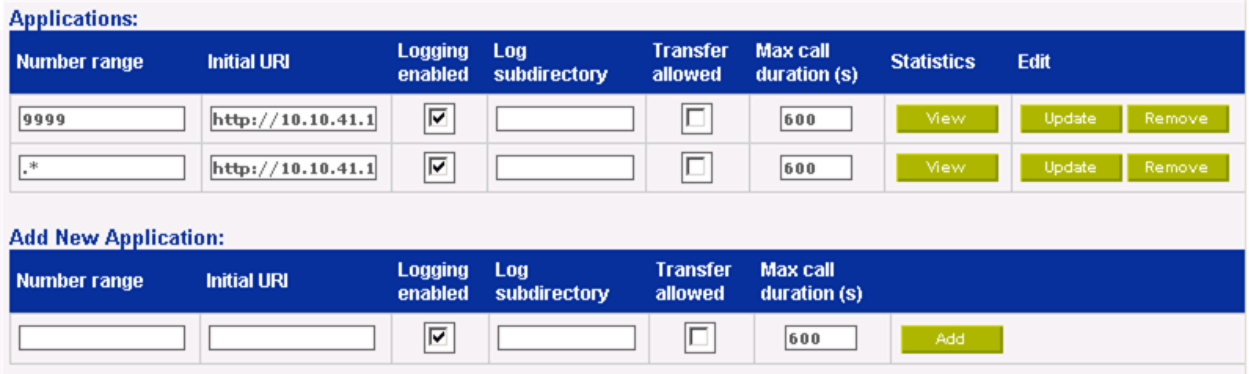
Adding a VoiceXML application

To add a VoiceXML application to the Vision Console, follow these steps:

Step	Action
1	Access the Vision Console, as described in Accessing the Dialogic Vision Console.
2	Click **VoiceXML applications** on the Provisioning menu. The VoiceXML application configuration page appears. To define a default application for all incoming calls, use the .% pattern. If no other matches are found for an incoming call, the initial URI associated with the .% pattern is used.

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>3</td><td>In the Add new application box, enter the following information for the new application:
<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>Number range</td><td>Number range that maps to the application. The number range can be a combination of digits and wildcards.
If the dialed number of an incoming call matches the number ranges of multiple applications, the call is matched to the most specific number range.
For information about the syntax used for specifying number ranges, see Pattern matching syntax.</td></tr>
<tr><td>Initial URI</td><td>Initial URI to use for a call, based on the number range of the dialed number (DNIS).
If the Programmable Media Platform receives a call whose DNIS does not match the number range for any of the configured applications, the caller is redirected to the URI specified in the DefaultInitialURI setting of the VoiceXML Interpreter configuration file. For more information, see the Dialogic® Vision™ VoiceXML Administration Manual.</td></tr>
<tr><td>Logging enabled</td><td>If checked, logging is per□ormed for the application.</td></tr>
<tr><td>Log subdirectory</td><td>Subdirectory to which application logs are written, if logging is enabled.</td></tr>
<tr><td>Transfer allowed</td><td>If checked, call transfers are allowed for the application.</td></tr>
<tr><td>Max call duration</td><td>Maximum call duration allowed for the application.</td></tr>
</table></td></tr>
<tr><td>4</td><td>Click Add.
The Vision Console adds the new definition to the top of the definition list.</td></tr>
</table>

The following example shows two defined applications. The first application matches incoming calls that have four-digit numbers. The second application matches all incoming calls that do not have four-digit numbers.



Removing a VoiceXML application

To remove an application from the Vision Console:

Step	Action
1	Access the Vision Console, as described in Accessing the Dialogic Vision Console.
2	Click **VoiceXML applications** on the Provisioning menu. The VoiceXML application configuration page appears.
3	Locate the row that contains the application you want to remove, and click **Remove.**

Modifying a VoiceXML application definition

To modify a VoiceXML application definition, follow these steps:

Step	Action
1	Access the Vision Console, as described in Accessing the Dialogic Vision Console.
2	Click **VoiceXML applications** on the Provisioning menu. The VoiceXML application configuration page appears.
3	Add the application to the Vision Console.
4	Fill in appropriate information for the application. For a description of the entry fields, see Adding a VoiceXML application to the Vision Console.
5	Click **Update** to save the changes.

Pattern matching syntax

The following table describes the syntax used to specify patterns for matching the dialed number (DNIS) for inbound calls:

Symbol	Description
. (period)	Wildcard that matches any single digit. For example, 123.... matches any dialed string beginning with 123, plus exactly four additional digits.
[ ]	Range of digits. A consecutive range is indicated with a hyphen (-); for example, [5-7]. A non-consecutive range is indicated with a comma (,); for example, [5,8]. Hyphens and commas can be used in combination; for example, [5-7,9]. **Note:** Only single-digit ranges are supported. For example, [98-102] is invalid.
( )	A pattern; for example, 408(555). Used in conjunction with the symbol ?, %, or +.
?	Preceding digit occurred zero or one time.
%	Preceding digit occurred zero or more times. This is similar to the asterisk (*) used in regular expression.
+	Preceding digit occurred one or more times.

The following table provides examples of destination patterns and how they are interpreted:

Pattern	Translation
408555.+	408555, followed by one or more wildcard digits. Indicates the string must contain at least 7 digits starting with 408555.
408555.%	408555, followed by zero or more wildcard digits. Indicates the string must contain at least 408555.
408555+	40855, followed by 5 repeated one or more times.
408555%	40855, followed by 5 repeated zero or more times. Any explicitly matching digit before the % symbol is not stripped off.
408555?	40855, followed by 5 repeated zero or one time. Any explicitly matching digit before the ? symbol is not stripped off.
40855[5-7].+	40855, followed by 5, 6, or 7, plus any digit repeated one or more times.
40855[5-7].%	40855, followed by 5, 6, or 7, plus any digit repeated zero or more times.
40855[5-7]+1234	40855, followed by 5, 6, or 7 repeated one or more times, followed by 1234.

Pattern	Translation
408(555)+1234	408, followed by 555, which may repeat one or more times, followed by 1234.

Managing CCXML applications

Use the Dialogic® Vision™ Console to specify the following properties for individual CCXML applications:

- A number range that maps to the CCXML application. Calls within the specified range are processed by the application.
- The initial URI to use for an incoming call, based on the specified number range.
- The dialog server that processes the initial URI.
- The outbound routes that the CCXML application uses for the PSTN calls that it creates.

The Vision Console adds this information to the CCXML application definition file (*ccxmlappcfg.xml*).

Note: To specify properties for an application, you must first add the application to the Vision Console.

This topic describes:

- Default CCXML application
- Adding a CCXML application definition
- Removing a CCXML application definition
- Modifying a CCXML application definition
- The pattern matching syntax used to match the dialed numbers of inbound calls and to define outbound routes

Default CCXML application

The Programmable Media Platform is shipped with a default CCXML application called *inbound.ccxml*, which is defined on the Vision Console as a custom application. This application matches all incoming calls and routes outbound calls to route 0. With route 0, the route is chosen automatically according to the associated incoming call. By default, all incoming calls are handled by dialog server 127.0.0.1:5070.

If you need additional routing functionality, you can do any of the following:

- Redefine *inbound.ccxml* to the Vision Console.
- Create one or more new CCXML applications, and define those applications as custom applications on the Vision Console.
- Access the gateway routing table functionality by creating one or more server applications. For information, see the *Dialogic® Vision™ 1000 Video Gateway Administration Manual*.

Adding a CCXML application definition

To add a CCXML application to the Vision Console, follow these steps:

Step	Action
1	Access the Vision Console, as described in Accessing the Dialogic Vision Console.
2	Click **CCXML applications** on the Provisioning menu. The CCXML application configuration page appears. By default, there is one CCXML application called *inbound.ccxml* defined for the Programmable Media Platform. This application is defined as a custom application. Gateway applications are used for routing when the Call Server is used as a gateway. For more information, see the *Dialogic® Vision™ 1000 Video Gateway Administration Manual*.

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>3</td><td>To add a new CCXML application to the Vision Console, click New in the Custom applications section, and enter the following information:
<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>Number range</td><td>Number range associated with the CCXML application specified by the Initial URI value. The number range can be a combination of numbers, alphabetic characters, and wildcards.

The Programmable Media Platform checks if the dialed number of an incoming call matches this. If it finds a match, the corresponding CCXML application is executed.

If the dialed number matches the number range of multiple applications, the call is matched to the most specific number range.

For more information, see Pattern matching syntax.</td></tr>
<tr><td>Initial URI</td><td>Initial URI to use for an incoming call, based on the number range of the dialed number.</td></tr>
<tr><td>Dialog servers</td><td>List of dialog servers for the application. Separate each dialog server with a comma.</td></tr>
<tr><td>Outbound routes</td><td>List of outbound routes for the calls that the CCXML application creates. Separate each route with a comma.

PSTN routes

For a PSTN route, the syntax is:

route-Route_Number[Priority]

where:

• Route_Number identifies a PSTN route, as defined in the telecom conf file.
• Priority specifies the priority level for load balancing over the telecom routes.

If Route_Number is set to 0, the route is chosen automatically according to the associated incoming call.

SIP routes

For a SIP route, the syntax is:

IP_Address:Port[Priority]

where:

• IP_Address:Port is the IP address and port of a SIP route.
• Priority the priority level for load balancing over SIP routes.

For both PSTN and SIP routes, Priority is</td></tr>
</table>
</td></tr>
</table>

Step	Action
4	Click **Apply**. The Vision Console adds the new definition to the top of the definition list in the Custom applications section.

Removing a CCXML application definition

To remove an application from the Vision Console, follow these steps:

Step	Action
1	Access the Vision Console, as described in Accessing the Dialogic Vision Console.
2	Click **CCXML applications** in the Provisioning menu. The CCXML application configuration page appears.
3	Locate the row that contains the application you want to remove, and click **Remove.** A confirmation message appears.
4	Click **OK** to remove the application.

Modifying a CCXML application definition

To modify a CCXML application definition, follow these steps:

Step	Action
1	Access the Vision Console, as described in Accessing the Dialogic Vision Console.
2	Click **CCXML applications** on the Provisioning menu. The CCXML application configuration page appears.
3	Click **Edit**. The Edit CCXML application page appears.
4	Modify the application as appropriate. For a description of the entry fields, see Adding a CCXML application definition.
5	Click **Apply** to save the changes.

CCXML application definition pattern matching syntax

The following table describes the syntax used to specify patterns for matching the dialed number (DNIS) for inbound calls. This is based on standard regular expression syntax.

Symbol	Description
. (period)	Wildcard that matches any single digit or character. For example, 123.... matches any dialed string beginning with 123, plus exactly four additional digits.
[ ]	Range of digits. A consecutive range is indicated with a hyphen (-); for example, [5-7]. A non-consecutive range is indicated with a comma (,); for example, [5,8]. Hyphens and commas can be used in combination; for example, [5-7,9]. **Note:** Only single-digit ranges are supported. For example, [98-102] is invalid.
( )	A pattern; for example, 408(555). Used in conjunction with the symbol ?, *, or +.
?	Preceding digit occurred zero or one time.
* or %	Preceding digit occurred zero or more times.
+	Preceding digit occurred one or more times.

The following table provides examples of destination patterns and how they are interpreted:

Pattern	Translation
408555.+	408555, followed by one or more wildcard digits. Indicates the string must contain at least 7 digits starting with 408555.
408555.*	408555, followed by zero or more wildcard digits. Indicates the string must contain at least 408555.
408555+	40855, followed by 5 repeated one or more times.
408555*	40855, followed by 5 repeated zero or more times. Any explicitly matching digit before the * (asterisk) is not stripped off.
408555?	40855, followed by 5 repeated zero or one time. Any explicitly matching digit before the ? (question mark) is not stripped off.
40855[5-7].+	40855, followed by 5, 6, or 7, plus any digit repeated one or more times.
40855[5-7].*	40855, followed by 5, 6, or 7, plus any digit repeated zero or more times.

Pattern	Translation
40855[5-7]+1234	40855, followed by 5, 6, or 7 repeated one or more times, followed by 1234.
408(555)+1234	408, followed by 555, which may repeat one or more times, followed by 1234.

Managing video transcoder resources

Video transcoding is the process of converting video media from one video codec type to another (for example, from H.264 to H.263) between two endpoints to suit the requirements of the device at each endpoint. Transcoding involves decoding and encoding of each frame of a video stream.

Video transrating adjusts the number of video frames per second (and bitrate of the video) between two endpoints to suit the requirements of the device at each endpoint.

Image resizing converts video from one image size to another (for example, from CIF to QCIF) between two endpoints to suit the requirements of the device at each endpoint.

In this document, the term video transcoding or video transcoder encompasses video transcoding, video transrating, and image resizing.

For a list of video codecs supported by the video transcoder, see Media capabilities. For an overview of the ways in which a video transcoder can be deployed, see Models with Video Transcoders.

The procedures for managing video transcoder resources include:

- Configuring a video transcoder system
- Defining video transcoder resources for the Programmable Media Platform
- Specifying video transcoding in a call leg
- Video transcoder logging

Configuring a video transcoder system

Follow these steps to configure each video transcoder system in your environment:

Step	Action
1	Access the Vision Console for your video transcoder system.

Step	Action
2	Click **Video Transcoder** on the Configuration menu, and the Video Transcoder page is displayed with default values based on your license. • Modify the number of full-duplex video transcoder channels that are available for this system as needed. The default value is the maximum number of licensed channels. The number of channels configured for use may be less than or equal to the maximum allowed by the license. To restrict usage, you can specify a number that is less than the maximum number of licensed video transcoder resources. • Modify the high water mark for CPU usage in percentage as needed. If this threshold is reached, the system issues an SNMP notification. • Modify the low water mark for CPU usage in percentage as needed. If this threshold is reached, the system issues an SNMP notification. • Modify the upper limit of high water mark for CPU usage in percentage. If this threshold is reached, the system issues an SNMP notification and begins to reject calls. • Modify the lower limit of low water mark for CPU usage in percentage. If this threshold is reached, the system issues an SNMP notification and begins to accept calls. • Click **Submit** to apply the changes.
3	Click **Services** on the Operations menu. On the Services page, start or restart the Video Transcoder service to apply the new configuration.

Defining video transcoder resources for the Programmable Media Platform

This procedure applies to a Programmable Media Platform that is mated with one or more video transcoder systems, which may be external or co-located.

To define video transcoder resources for a Programmable Media Platform, follow these steps:

Step	Action
1	Access the Vision Console for your Programmable Media Platform, as described in Accessing the Dialogic Vision Console.
2	Click **Resources** on the Configuration menu, and the Resource Configuration page is displayed. • In the Global resources group, enable video transcoding. This global flag indicates whether video transcoder resources are available for the Programmable Media Platform. • Click **Submit** to apply the change.
3	Click **Services** on the Operations menu. On the Services page, start or restart the Call Server service to apply the new configuration.

Step	Action
4	Click **Video transcoder resources** on the Provisioning menu, and the Video transcoder resource configuration page is displayed. • Specify the IP address of a video transcoder system to be attached to the Programmable Media Platform and click **Add video transcoder**. The IP address is listed in the table as well as the system name and number of channels. The Media Platform determines if video transcoder services are running on that system and displays a message accordingly. • Specify the IP address of each additional video transcoder system to be deployed, if any, and click **Add video transcoder**. • To remove a video transcoder system, click **Remove**. • Click **Submit** to apply the changes. After you have updated values on the Provisioning menu, you do not have to restart the Call Server service to apply the new configuration.

Specifying video transcoding in a call leg

If you have enabled video transcoding for the Programmable Media Platform on the Resources page in the Configuration menu, transcoding will be used only when incompatible video codec characteristics are detected by the Programmable Media Platform. If needed, you can force every call to use video transcoding through the routing profile.

To force video transcoding for every call, create or edit the routing profile as follows:

Step	Action
1	Access the Vision Console for your Programmable Media Platform, as described in Accessing the Dialogic Vision Console.
2	Click **Gateway profiles** on the Provisioning menu, and the Routing profiles configuration page is displayed with a default profile.
3	In the General section, edit the default profile (or select the desired profile to be edited from the Current profile drop-down list) and select **Force** in the Video transcoding field.
4	Click **Submit** to apply the changes.
5	Click **Gateway routes** in the Provisioning menu. The Call routing table page is displayed. Verify that the updated profile is associated the appropriate routing entry.

When **Force** is specified in the profile and associated with a route, video transcoding is inserted in the video path regardless of the negotiated video codec on either side. When **Dynamic** is specified, video transcoding only applies to calls with different video codec characteristics such as a different codec or a different picture frame. For more information on routing profiles and routing tables, see the *Dialogic® Vision™ 1000 Video Gateway Administration Manual.*

Benefits of always enabling video transcoding in the path can include the following:

- Enables the server to respond to a VFU request without relying on the remote endpoint.
- For MPEG-4 codec in particular, ensures that the DCI information will not change in the middle of the session between a 3G endpoint and an RTP endpoint.

Video transcoder logging

Video transcoder log files are created in the */opt/nms/video/logs* directory on the video transcoder system.

Note: Video transcoder log files are intended for use by Dialogic Technical Services and Support.

The default logging level is ERROR. The logging level is configurable on the Maintenance page, Operations menu of the Vision Console. The logging levels are identical to the Call Server system log levels described in Logging levels.

The following information is provided to help you manage your system requirements according to the number of video transcoder resources in use in your environment:

- The maximum size of each log file is 10 MB.
- When an individual log file reaches this maximum size, it is rolled over and a new file is created. The maximum rollover files is five. At any point, there may be six files for each type of log file: one active log file and five rollover log files.
- When video transcoder services are stopped and started, the *logs* directory is rolled over. For example, the first time after installation, all logs are created in the *logs* directory. If the video transcoder services are stopped and restarted, *logs.1* directory is created which stores past log files. Current logging occurs in the *logs* directory. The logs directory rollover value is 2. At any point, there may be three logs directories: *logs*, *logs.1*, and *logs.2*.

For example, in a system that uses 60 video transcoder channels, the maximum number of log files that reside in the *logs* directory may be up to 1632 files (272 active log files along with 5 rollover log files for each). The maximum combined size of these log files may be up to about 16 GB (1632 files at 10 MB each).

Video call completion to voice service

The Video Call Completion to Voice (VCCV) service allows a 3G video call to be connected as a voice-only call when the called party is not able to receive video calls. This service may be useful in the following call scenarios:

- Called party is not a 3G subscriber.
- Called party is a 3G subscriber but is out of 3G coverage.
- Called party is a 3G subscriber but is in a busy, no answer, or switched-off condition.
- Called party is roaming in a network that does not support video calls.
- Called party has no subscription to video calls.

The service terminates the initial video call and starts a separate audio call to the called party. During the audio conversation, the service streams application-defined video content to the 3G calling party.

VCCV provides an interactive option which allows the calling party to decide whether to proceed with an audio call or whether to disconnect the original call. After the prompt is played, if the calling party doesn't respond, the call will be disconnected.

Note: The interactive option of VCCV requires Programmable Media Platform licenses. By default, the interactive option is not enabled.

VCCV provides an option for video fallback to a voice call if a 3G video call fails to connect. This fallback is triggered by configurable cause codes. If a video call fails to connect due to a cause code specified in the list, the Vision Server will execute the VCCV or interactive VCCV logic.

VCCV provides early media support. If early media is enabled, media begins to flow in both directions before the 3G video call is connected. In an SS7 network, this means that the 3G-324M negotiation starts after the address complete message (ACM) rather than the answer message (ANM) is received from the Call Server.

Call logic

At a high level, the call logic for Video Call Completion to Voice (VCCV) is as follows:

1. A user makes a video call from a 3G handset to a party that cannot receive video calls.
2. The network determines that bearer capability is not supported and redirects the call to the Dialogic® Vision™ Server.
3. If the routing profile associated with the routing table entry specifies VCCV, the Vision Server then terminates the initial video call and places an audio call to the called party.

   If interactive VCCV is enabled, the calling party is presented with a menu that allows him to continue with an audio call or to disconnect the original call.
4. Without the early media option, after the called party answers the call, an audio announcement is played to the called party while 3G negotiation takes place. The call is typically connected after the audio announcement is complete. In some cases, depending on network traffic and when 3G negotiation is complete, the call may take a little longer to be connected.
5. If the early media option is enabled, the Vision Server begins to stream a video ringback file while the called party's handset is ringing. When the called party answers the phone, the server stops the video ringback file, and bridges the audio path between the two parties. No audio announcement is played to the called party.
6. A video background file is played to the calling party while the call is in progress.
7. The call is complete and both sides disconnect.

Note: To play network announcements, you must have announcement port capacity; see Capacity Upgrade.

Using the service

To use Video Call Completion to Voice (VCCV), follow these steps:

Step	Action
1	Click **CCXML applications** in the Provisioning menu. If it is not already listed, add the *gateway.ccxml* application. For more information, see Managing CCXML applications. The default *gateway.ccxml* script contains VCCV application logic.

Step	Action
2	For interactive VCCV, create a new Gateway Application or edit an existing one in CCXML applications in the Provisioning menu. Set the Initial URI field to *file:///opt/nms/vx/callserver/www/ccxml/i-gateway.ccxml.* For more information, see Managing CCXML applications. The *i-gateway.ccxml* script contains interactive VCCV application logic with a preset menu. To create a custom menu, you will need to update the VoiceXML application located in the */opt/nms/vx/vxmlinterpreter/www/ivccv/* directory.
3	Click **Gateway routes** in the Provisioning menu. Edit the default profile or create a new profile and set the Outbound call mode to **voice+** (for standard VCCV or for interactive VCCV) in the General section of the Routing profile configuration page. Select the Video Fallback to Audio option as needed in the General section, and specify the cause codes that will trigger this option in the PSTN section. Select other options as needed, such as early media, in the VCCV section of the page. For more information, see *Dialogic® Vision™ 1000 Video Gateway Administration Manual*. You do not need to restart the Call Server after making changes in the Provisioning menu.
4	Assign this profile to the desired route in the routing table. For more information, see the *Dialogic® Vision™ 1000 Video Gateway Administration Manual.*

Configuring streaming-only media server applications

In some use cases, you may want a media server to send high-quality 3G video through the Programmable Media Platform to a 3G handset without receiving media in return. Doing so helps to conserve video transcoder resources.

To support this use case, the Programmable Media Platform application must specify **a=sendonly** in SDP media lines and must comply with RFC 3264, An Offer/Answer Model with SDP.

You must also set these two parameters in the *vxmlinterpreter.conf* file to **sendonly**:

- com.vision.miosip.media.VideoCallPreferredVideoDirection
- com.vision.miosip.media.VideoCallPreferredAudioDirection

Working with Ethernet Redundancy

Ethernet redundancy refers to the ability of the Dialogic® Vision™ Server to reach the network it is connected to through redundant network interfaces. It allows devices on the network to reach the server and vice versa if one of the connections to the server fails.

In addition, the Vision Server supports addressing multiple independent IP networks. This is achieved by configuring multiple IP addresses for the same network adapter on the server. You can configure up to seven predefined IP networks on the server:

- SIP
- RTP
- Circuit-switched signaling
- NbUP
- Billing
- OA&M
- Redundant circuit-switched signaling

You can configure a different IP address for SIP, RTP, and so on. In order to properly separate the traffic, you need to assign each predefined IP network to a specific traffic type.

Since the Vision Server has a limited number of network interfaces, it is also possible to enable VLAN tagging on the server's configured network interfaces to provide proper network isolation.

Ethernet redundancy is configurable through the Dialogic® Vision™ Console. You can configure Ethernet bonding; create interface aliases; enable on-host and on-board VLAN tagging; and assign traffic types to configured interfaces.

The following topics provide more information about Ethernet redundancy:

- Ethernet redundancy concepts
- Configuring the SIP network
- Configuring the RTP network
- Configuring the Circuit-Switched Signaling network
- Configuring the NbUP network
- Configuring the Billing network
- Configuring the OA&M network
- Configuring the Signaling Redundant network

See Network redundancy and the network monitor service for related information.

Ethernet redundancy concepts

Ethernet bonding is used to link two physical Ethernet ports on the host in a redundant manner. The Vision Server implements bonding in an active-backup configuration. Only one port in the bond is the active port; the other port serves as a backup if the active port fails. Bonding is enabled by assigning a bond interface to two physical interfaces.

An interface alias is used to link multiple IP addresses to a single physical network interface. This is also known as multi-homing. An interface alias is enabled by assigning multiple IP addresses to a single interface. Up to 16 aliases are supported per interface.

A virtual LAN, or VLAN, is a group of hosts with a common set of requirements that communicate as if they were attached to the same broadcast domain regardless of their physical location. VLAN is used to enable network splitting or network isolation. Assigning a VLAN ID (also called VLAN tagging) to each virtual interface allows each interface to be isolated. The Vision Server follows the IEEE 802.1Q standard for VLAN tagging.

Configuring the SIP network

To configure the SIP network for Ethernet redundancy, follow these steps:

Step	Action
1	Determine the Ethernet redundancy requirements for your environment. See SIP network for more information.
2	If you haven't already, access the Vision Console as described in Accessing the Dialogic Vision Console.
3	Click on **Host IP information** in the Configuration menu. The Host IP information page is displayed.
4	Define an interface for the SIP network and assign the Signaling traffic type to this interface. You can also configure Ethernet bonding, create interface aliases, enable VLAN tagging, and define IP routes if needed. See Host IP information for field descriptions. Click **Submit**.
5	Continue to configure other predefined IP networks in your system.
6	After you have finished configuring all predefined IP networks, restart services. Click on **Services** in the Operations menu, and then click **Restart all**. Once the status of all platform services is STARTED, you can proceed.

Configuring the RTP network

If you do not use a separate network for RTP traffic, see RTP parameters section to configure the media board network interface.

To configure the RTP network for Ethernet redundancy, follow these steps:

Step	Action
1	Determine the Ethernet redundancy requirements for your environment. See RTP network for more information.
2	If you haven't already, access the Vision Console as described in Accessing the Dialogic Vision Console.
3	Click on **Host IP information** in the Configuration menu. The Host IP information page is displayed.
4	Define an interface for the RTP network and assign the Media traffic type to this interface. You can also configure Ethernet bonding, create interface aliases, enable VLAN tagging, and define IP routes if needed. See Host IP information for field descriptions. Click **Submit**.

Step	Action
5	Click on **RTP** in the Configuration menu. The RTP parameters page is displayed.
6	Define an interface on the media board and configure it to be on the same network as the RTP network you created in Step 4. Assign the RTP traffic type to this interface. Configure the redundant status on the second interface as needed; it would become redundant to the first interface. See RTP parameters for field descriptions. Click **Submit**.
7	Continue to configure other predefined IP networks in your system.
8	After you have finished configuring all predefined IP networks, restart services. Click on **Services** in the Operations menu, and then click **Restart all**. Once the status of all platform services is STARTED, you can proceed.

Configuring the Circuit-Switched Signaling network

If you do not require SIGTRAN or BICC traffic in your system, disregard this section.

To configure the circuit-switched signaling network, follow these steps:

Step	Action
1	Determine the Ethernet redundancy requirements for your environment. See Circuit-switched signaling network for more information.
2	If you haven't already, access the Vision Console as described in Accessing the Dialogic Vision Console.
3	Click on **Signaling Server** in the Configuration menu. The Signaling Server page is displayed.
4	Specify IP address information for the Signaling (TX) board. To get to the TX Board fields, set Transport to SIGTRAN or set ISUP switch type to ITUBICC or ANSIBICC. See Signaling Server for field descriptions. Click **Submit**.
5	Continue to configure other predefined IP networks in your system.
6	After you have finished configuring the predefined IP networks, restart services. Click on **Services** in the Operations menu, and then click **Restart all**. Once the status of all platform services is STARTED, you can proceed.

Configuring the NbUP network

If you do not use a separate network for NbUP traffic, disregard this section.

To configure the NbUP network for Ethernet redundancy, follow these steps:

Step	Action
1	Determine the Ethernet redundancy requirements for your environment. See NbUP network for more information.
2	If you haven't already, access the Vision Console as described in Accessing the Dialogic Vision Console.
3	Click on **RTP** in the Configuration menu. The RTP parameters page is displayed.
4	Define an interface for the NbUP network and assign the NbUP traffic type to this interface. Define the IP route if needed. See RTP parameters for field descriptions. Click **Submit**.
5	Continue to configure other predefined IP networks in your system.
6	After you have finished configuring all predefined IP networks, restart services. Click on **Services** in the Operations menu, and then click **Restart all**. Once the status of all platform services is STARTED, you can proceed.

Configuring the Billing network

If you do not use a separate network for Billing traffic, disregard this section.

To configure the Billing network for Ethernet redundancy, follow these steps:

Step	Action
1	Determine the Ethernet redundancy requirements for your environment. See Billing network for more information.
2	If you haven't already, access the Vision Console as described in Accessing the Dialogic Vision Console.
3	Click on **Host IP information** in the Configuration menu. The Host IP information page is displayed.
4	Assign the Billing traffic to the appropriate network interface. Define the IP route if needed. See Host IP information for field descriptions. Click **Submit**.
5	Continue to configure other predefined IP networks in your system.
6	After you have finished configuring the predefined IP networks, restart services. Click on **Services** in the Operations menu, and then click **Restart all**. Once the status of all platform services is STARTED, you can proceed.

Configuring the OA&M network

If you do not use a separate network for OA&M traffic, disregard this section.

To configure the OA&M network for Ethernet redundancy, follow these steps:

Step	Action
1	Determine the Ethernet redundancy requirements for your environment. See OA&M network for more information.
2	If you haven't already, access the Vision Console as described in Accessing the Dialogic Vision Console.
3	Click on **Host IP information** in the Configuration menu. The Host IP information page is displayed.
4	Assign the OA&M traffic to the appropriate network interface. Define the IP route if needed. See Host IP information for field descriptions. Click **Submit**.
5	Continue to configure other predefined IP networks in your system.
6	After you have finished configuring the predefined IP networks, restart services. Click on **Services** in the Operations menu, and then click **Restart all**. Once the status of all platform services is STARTED, you can proceed.

Configuring the Signaling Redundant network

If you do not use a separate network for circuit-switched signaling redundancy, see the Signaling Server section to configure the redundant network interface of the signaling board.

To configure the Signaling Redundant network for Ethernet redundancy, follow these steps:

Step	Action
1	Determine the Ethernet redundancy requirements for your environment. See Signaling Redundant network for more information.
2	If you haven't already, access the Vision Console as described in Accessing the Dialogic Vision Console.
3	Click on **Signaling Server** in the Configuration menu. The Signaling server page is displayed.
4	Fill in the fields as appropriate. For example, configure the redundant status for the signaling server as appropriate. See Signaling Server for field descriptions. Click **Submit**.

Step	Action
5	Continue to configure other predefined IP networks in your system.
6	After you have finished configuring all predefined IP networks, restart services. Click on **Services** in the Operations menu, and then click **Restart all**. Once the status of all platform services is STARTED, you can proceed.

Network redundancy and the network monitor service

The Programmable Media Platform supports network redundancy and provides a network monitor service. The primary role of this service is to monitor specified IP addresses and perform a failover if these addresses become unavailable. This service also ensures that redundant network interface pairs are synchronized across the system, so that the host and the media boards in the system can exchange RTP traffic as needed.

Failover is defined as an automatic fallback to a redundant backup device when the primary device fails. **Switchover** is a failover that is triggered by manual intervention.

You can enable network redundancy and configure the following parameters in the Vision Console:

- Monitored interfaces
- Monitored networks and IP addresses
- Monitoring frequency in milliseconds
- Monitoring timeout

The network monitor service considers the first interface of a redundant pair as the primary interface, and the second one as the backup.

If the network monitor service detects that a failover occurred in one interface group, it will force a failover for all interface groups, so that all active interfaces operate from the same Ethernet switch.

Configuring the network monitor service

Follow these steps to use network redundancy and configure the network monitor service:

1. Determine Ethernet redundancy requirements for your environment. See Ethernet redundancy configuration information.
2. Configure your environment for Ethernet redundancy. See Working with Ethernet redundancy.
3. Determine the network monitor service requirements for your environment. See Network monitor configuration information.
4. Access the Vision Console, as described in Accessing the Dialogic Vision Console.
5. Click on **Network Redundancy** in the Configuration menu. The Network Redundancy Configuration page is displayed.
6. Fill in the fields as appropriate. For example, enable network redundancy, enable monitoring for all available interfaces, and configure monitored networks. See Network redundancy configuration for parameter descriptions. Click **Submit**.

7. After you have finished configuring network redundancy, restart services. Click on **Services** in the Operations menu, and then click **Restart all**. Once the status of all gateway services is STARTED, you can proceed.

Out-of-band management

Out-of-band management allows you to monitor and manage the Vision Server remotely using a dedicated management channel, regardless of whether the server is powered on. A remote management interface is included with the Dialogic® Vision™ AQR1U Server model. Other models do not support this feature.

For a current list of models that support this feature, see the *readme* file for the release.

Using the remote management interface

The remote management interface uses the eth3 interface on the Vision Server and has the following default IP network configuration:

- IP address: 192.168.0.2
- Subnet mask: 255.255.255.0
- Gateway: none

The eth3 interface is shared between the operating system and the management module. The interface has two MAC addresses and two IP addresses, but only one cable is connected to it. See the hardware installation guide for the Dialogic® Vision™ AQR1U Server for details on the Ethernet interfaces.

To access the remote management interface, follow these steps:

Step	Action
1	Assign IP address 192.168.0.100 to the computer that will access the remote management interface.
2	Connect this computer to eth3 on the Vision Server either directly using a crossover cable, or connect through a standalone Ethernet hub or switch.
3	Enter the following URL from a Microsoft® Internet Explorer or Firefox browser on the computer: http://192.168.0.2 You are redirected to secure HTTP (HTTPS) connection. A message related to the web site's security is displayed. You can click to continue to the web site, or you can install a security certificate on the system. For information, see Installing a security certificate.
4	Log into the remote management interface using the following information: • User: vision-root • Password: vision The System Information page is displayed.

Step	Action
5	Click **Remote Control** in the top menu bar. The Remote Control page is displayed with two options.
6	To launch the redirection console viewer, click **Console Redirection**. **Note:** You will need to install the Java Runtime Environment to use this option.
7	To see the server power status and perform power control functions, click **Power Control**. The following power control functions are available: • Reset Server • Power Off Server - Immediate • Power Off Server - Orderly Shutdown • Power On Server • Power Cycle Server

Managing Vision Nodes

For installations where density requires multiple Vision Servers, you have the ability to manage a group of two or more Vision Servers as one logical unit. This group of Vision Servers is referred to as a **Vision node**.

A Vision node can consist of a combination of Video Gateways and Programmable Media Platforms, including models with video transcoders.

For example, a Vision node can consist of the following:

- Two Programmable Media Platforms with a Signaling Server and one media board, where each Programmable Media Platform provides 120 video ports
- Two Programmable Media Platforms with two media boards, where each Programmable Media Platform provides 240 video ports

The resulting Vision node contains redundant SS7 signaling and 720 video ports.

Vision nodes are configurable from the Configuration menu of the Vision Console.

Vision node concepts

A Vision Server that is part of a Vision node is referred to as **node member**.

Each server of a Vision node can be used to manage the full node. A node member that is used to access and manage a node is referred to as a **node manager**. A node manager can support multiple console clients.

Vision node guidelines

Follow these guidelines when creating and working with a Vision node:

- A Vision Server can only be part of one node.
- A node can have at most two Signaling Servers.
- Each node member is configured to be aware of all the members in the node.

- Members of the same node are assumed to be physically co-located; that is, on the same Ethernet switch or segment. This is to avoid delays when synchronizing information across node members.
- Node members are not required to have the same hardware configuration. For example, some members can provide SS7 connectivity while other members only provide media processing.
- A node can be managed from any member of the node. The member used by a web client to access a node's console interface is referred to as its node manager.
- Connecting to a node manager's Vision Console provides access to the full node, assuming all node members are available.
- A node member can be temporarily excluded from the node (that is, disabled). This allows a node member to remain in the node configuration even if it is physically unavailable.

Defining a node

Follow these instructions to define a node:

1. Determine the node requirements for your environment, which includes determining an IP address for each server. See Node configuration information.
2. Access the Vision Console using one of the servers in the node. This server is considered the node manager.
3. Access the Node definition page by clicking on **Node Definition** in the Configuration menu.
4. Enter the node name. Define the node using the list of IP addresses. Click **Add** to add a node member.
5. After you have completed the node definition, click **Deploy**. This action causes the node manager to propagate the node definition to all node members. Global configuration changes to a node and its members will be successfully applied only if all node members are available. If a node member is not available, the action is not successfully completed and an error is reported.
6. After the Vision node is successfully created, you can configure and provision each node member from the Vision Console of the node manager.
7. After you have finished configuring all node members, restart services. Click on **Services** in the Operations menu, and then click **Restart all**. Once the status of all gateway services is STARTED, you can proceed.

Disabling or enabling a node member

You may need to disable a node member if a node member becomes unavailable.

You can enable or disable a node member from the Node definition page in the Configuration menu. Click the Enabled check box to enable a node. Leave blank to disable a node.

Changes can only be applied if all enabled node members are available to ensure configuration integrity across the node. Member-specific configuration is not accepted for a disabled member.

This operation does not require a services restart.

Removing a node member

You can remove a node member from the Node definition page in the Configuration menu. Click **Remove** to remove a node member and apply the change.

Note: You should only remove a node member as a definitive operation; otherwise, you should disable the node member.

Removing a node member may result in the automatic renaming of some node members so that members are numbered sequentially. For example, if you remove NodeA-2 from NodeA which has three members, NodeA-3 will subsequently be renamed to NodeA-2. NodeA-1 will not change.

This operation requires you to restart services on affected node members.

Upgrading node capacity

You can upgrade node capacity from the Node definition page in the Configuration menu. Click **Add** to add a node member and apply the change.

This operation requires you to restart services on the new node member.

Using SIP load balancing

SIP load balancing allows you to distribute and balance the amount of SIP service network traffic among available Vision Servers for performance scalability and high availability.

This feature enables a group of Vision Servers to be reachable through the same SIP IP address by introducing IP virtualization service.

SIP load balancing is enabled and configured from the Configuration menu of the Vision Console. Logging information is available from the Monitoring menu.

How SIP load balancing works

The following steps provide an overview of how SIP load balancing works:

1. A single SIP agent allows the SIP network to contact the Vision Servers in the system environment. The SIP agent is available on two Vision Servers to ensure availability. The SIP network reaches the SIP load balancer using a shared virtual IP address.
2. The SIP load balancer handles each incoming SIP call by redirecting it to an appropriate Vision Server. The SIP agent uses the *302 Moved Temporarily* response message to perform this action. If no Vision Server is available to answer the call, the SIP load balancer returns *503 Service Unavailable*.
3. The SIP load balancer periodically polls the Vision Servers to determine their availability. The SIP load balancer chooses the Vision Server with the higher free capacity as the redirection target.
4. The SIP load balancer assumes that all Vision Servers are able to handle every incoming call; that is, all Vision Servers have the same gateway routes defined.
5. You can monitor the state of the configured virtual IP addresses from the Monitoring menu. If needed, you can request a switchover for the listed virtual IP addresses.

Configuring SIP load balancing

The following steps describe how to configure and use SIP load balancing:

1. Determine SIP load balancing requirements and network redundancy requirements for your environment. See SIP load balancing configuration information, Routes

configuration information, and Network Monitor configuration information for more information.

2. Access the Vision Console as described in Accessing the Dialogic Vision Console. For parameter descriptions, see the Configuration menu parameters section.
3. Access the SIP parameters page by clicking on **SIP** in the Configuration menu.

   Click the **Enabled** check box next to SIP load balancing to enable this feature. The Vision Console automatically updates the SIP ports and places the SIP load balancer on port 5060. You should review these port updates to be sure they suit your needs and make changes if needed.

   Configure the SIP load balancing server information, including server names and IP addresses.
4. Access the Host IP information page by clicking on **Host IP information** in the Configuration menu. Specify which SIP interface alias will serve as the virtual IP by clicking the VIP check box. For configuration steps, see Configuring the SIP network.
5. Access the Network redundancy page by clicking on **Network redundancy** in the Configuration menu. In order to define an external address to ping to verify network availability, you must assign each interface associated with a virtual IP address to a previously defined monitored network. For more information, see Network redundancy and the network monitor service.
6. After you have finished configuring SIP load balancing, restart services. Click on Services in the Operations menu, and then click **Restart all**. Once the status of all gateway services is STARTED, you can proceed.
7. To view status information for the monitored virtual IP addresses, click on **Network monitoring** in the Monitoring menu. You can request a virtual IP address switchover on this page if needed.

6. Developing VoiceXML applications

Overview of developing VoiceXML applications

Use VoiceXML 2.1 to develop speech and video applications that interact with the Programmable Media Platform.

This section describes how to use VoiceXML to:

- Manage media
- Develop an audio application
- Develop a video application
- Use the record utterance functionality

Managing media

VoiceXML applications have file:// access to local files and to files that reside on NFS shares. They also have HTTP:// access to remote media files on web servers.

The HTTP:// access works as follows:

- All media files are fetched from a web server over HTTP or HTTPS.
- The Programmable Media Platform reads the MIME type of the media file as presented by the web server. If the MIME type is not available, the Programmable Media Platform uses the filename extension to determine the media type of the file.
- The Programmable Media Platform implements and follows all caching parameters as presented in the HTTP headers for the media files.
- All announcement media files are available for playing as soon as they are installed in the correct location on the application server.
- Media files with a streamable format begin streaming to the bearer connection when the HTTP transfer starts. The media server does not wait for the entire media file to transfer.

Developing an audio application

With all models of the Programmable Media Platform, the application uses standard VoiceXML 2.1 elements to develop audio applications.

For information about standard VoiceXML elements, including their descriptions and syntax, see the *Dialogic® Vision™ VoiceXML Reference Manual*.

Developing a video application

The ISDN video, ISUP video, and IP-only video models of the Programmable Media Platform use VoiceXML extensions in the video application to play, record, and post video messages.

Playing a video message

Use the VoiceXML <audio> element to play a video message in a 3GP file.

```
<prompt>
   Message 1, received yesterday at 10:45 pm.
   <audio src="http://192.168.1.1/msg01.3gp"/>
   End of message.
</prompt>
```

Note: When a video clip finishes playing, the last frame remains on the screen.

Because video clips play synchronously within other audio markup, the mixing of audio and video clips is seamless. Video clips may or may not include audio.

For more information about the VoiceXML <audio> element, see the *Dialogic® Vision™ VoiceXML Reference Manual*.

The Programmable Media Platform supports multiple video codecs (H.263, MPEG-4, and H.264). The VoiceXML application must select a 3GP file that has the appropriate codec based on the negotiated codec for each call. To do so, the application can use the proprietary variable session.connection.videostreamtype in the file name or as part of the file path to select the file. For example:

```
<prompt>
   Message 1, received yesterday at 10:45 pm.
   Using session.connection.videostreamtype as part of the file name
   <audio expr="'http://192.168.1.1/msg01-'+session.connection.videostreamtype+'.3gp'"/>
   End of message.
</prompt>
or
<prompt>
   Message 1, received yesterday at 10:45 pm.
   Using session.connection.videostreamtype as part of the path
   <audio expr="'http://192.168.1.1/'+session.connection.videostreamtype+'/msg01.3gp'"/>
   End of message.
</prompt>
```

Recording and posting a video message

Use the VoiceXML <record> element to record a video clip. For example:

```
<record name="videomsg" beep="true" type="video/3gpp">
   <prompt timeout="5s">
      Record a video message after the beep.
   </prompt>
</record>
```

Then use the <submit> element as a MIME body in an HTTP POST request to send the message to a web server. For example:

```
<field name="confirm">
<grammar type="application/srgs+xml" src="/grammars/boolean.grxml"/>
<prompt>
   Here is your recorded video message. <audio expr="videomsg"/>
</prompt>
<prompt>
   To keep it, say yes. To discard it, say no.
</prompt>
<filled>
   <if cond="confirm">
       <submit next="save_message.pl" enctype="multipart/form-data" method="post"
namelist="videomsg"/>
   </if>
   <clear/>
</filled>
/field>
```

Note: If the application allows multiple codecs, then every prompt must be encoded for each supported codec and saved to a separate 3GP file. For example, suppose the application supports H.263 and H.264, and it has one hello prompt. The application needs to create two 3GP files for the hello prompt: one for H.263 and one for H.264.

MPEG-4 limitation for 3G-324M handsets

When building video VoiceXML applications intended to be used by 3G-324M handsets, MPEG-4 has a known limitation. The MPEG-4 decoder configuration information (DCI) is

negotiated as part of the initial 3G-324M call setup, and can only be changed by closing and re-negotiating a new video channel on the 3GP leg of the call. This procedure is not widely supported by 3G handsets.

Because of this limitation, you should consider the following options:

Action	Options
Playing a video clip	Encode all MPEG-4 3GP files using identical parameters. Use the associated DCI to configure the mpeg4DecoderInfo parameter for the Call Server. For more information, see the *Dialogic® Vision™ Call Server Administration Manual*.
Recording and posting a video message	Do not use MPEG-4 for these actions, because the DCI used on the handset may be incompatible with that used by the Programmable Media Platform to play back the file to the user.

Using the record utterance functionality

To use the record utterance functionality in a VoiceXML application, follow these steps:

Step	Action

1	Set the "recordutterance" property in the VoiceXML script where the simultaneous recording and recognition will occur. For example:
	`:`
	`<property name="recordutterance" value="true"/>`
	`:`
	`<form id="myForm">`
	`<field name="myField">`
	`:`
	`</field>`
	`</form>`
	`:`

2	After recognition, you can reference the recognized utterance by using the shadow variable "lastResult$.recording".
	For example, the following sample script plays back the recognized utterance and submits it to an external server for storage:
	`:`
	`<filled>`
	`<prompt>You just said <value expr="lastResult$.recording"></prompt>`
	`<submit next="http://myserver.mydomain.com/persist utterance.jsp"`
	`method="post" namelist="lastResult$.recording"`
	`enctype="multipart/form-data"/>`
	`</filled>`
	`:`

7. Logging

Overview of Programmable Media Platform logging

The Programmable Media Platform logs error codes and alarms and can report statistics for incoming and outgoing calls. The following Programmable Media Platform components generate log files:

- Call Server
- VoiceXML Interpreter
- SSML Processor
- VoiceXML application
- SSML application

This section also discusses logging and performance.

Call Server system logs

The Call Server creates system logs that record information about the active Call Server process. The name format for the log is:

*callserver_**creationdate**_[**index**].log*

where:

- ***creationdate*** is the date the log file was created, formatted as local server time by default.
- ***index*** is an integer specifying the current incremented Call Server system log file. This value is reset daily and incremented when the configured maximum system log file size is reached or when the Call Server is restarted. Because the index value is a timestamp, indexes are ordered, but not necessarily consecutive.

For example, the following Call Server system log file was generated by the callserver process on 2008-09-15. It has an index value of 4:

```
callserver_2008_09-15_004.log
```

To view log files, use the Log files option on the Monitoring menu. For more information, see Log files.

This topic describes the following information about Call Server system logs:

- Logging levels
- Logging defaults
- Changing the logging level
- Changing other logging defaults
- Log file format

Logging levels

The following table describes the logging levels for Call Server system logs in decreasing severity and increasing verbosity order:

Logging level	Description
FATAL	Logs only critical errors.
ERROR	Logs all errors.
WARNING	Logs all errors and warnings.
INFO1	Logs information useful for first-level debugging and also logs some normal events.
INFO2 INFO3	Logs information useful for second-level debugging. Because this level generates abundant log information, Dialogic recommends that you set this level only at the request of Dialogic Technical Services and Support.
INFO4 INFO5	The most verbose option, which logs low-level detailed information. Because this level generates abundant log information, Dialogic recommends that you set this level only at the request of Dialogic Technical Services and Support.

Logging defaults

The default values for Call Server system logs are:

- The logging level is set to 1, which displays events encountered during typical Call Server operations.
- The log file is stored in the *vx/callserver/logs* directory.
- The maximum number of log files in the log directory is 50. If the log directory contains 50 log files, then the oldest log file is deleted when the 51st log file is added.
- The maximum size of a log file is 10 MB. If a log file reaches that size, a new log file is created.
- The date/time format for the log creation date and log entry date is in local server time.
- The index for the Call Server system log file is reset each day. The index is incremented when a new log file is started because the maximum system log file size was reached or the 51st log file was added to the log directory.

Changing the logging level

To change the Call Server system logging level, follow these steps:

Step	Action
1	Access the Vision Console, as described in Accessing the Vision Console.

Step	Action
2	Click **Maintenance** on the Operations menu. The Maintenance page appears.
3	Select the desired log level in the Log level field associated with the Call Server service, and click **Submit**.

Changing other logging defaults

The following table describes how to change the other logging defaults for the Call Server system log:

To change the...	Modify the...
Log file location	LogDir setting in the *callserver.conf* file.
Maximum number of log files in the specified directory	SystemLogFileMaxNum setting in the *callserver.conf* file.
Maximum size of the log file	SystemLogFileMaxSize setting in the *callserver.conf* file.
Date/time format of the log file	SystemLogTime setting in the *callserver.conf* file.

For more information, see the *Dialogic® Vision™ Call Server Administration Manual*.

Log file format

The format of each Call Server system log message is:

```
time-stamp [severity] [origin:code] [UID:threadID] (alarm) [message]
```

The following table describes the Call Server system log fields:

Field	Description
severity	Level of the log message. Valid values (in order of decreasing severity and increasing verbosity): • FATAL (a severe malfunction from which the server processor cannot recover) • ERROR • WARNING • INFO1 • INFO2 • INFO3 • INFO4 • INFO5 (highest/most verbose level of detail)

Field	Description
origin	Call Server component to which the log message refers.
code	Trace identifier of the message in the Call Server component to which the log message refers.
UID	Reserved for future use. The UID is represented by a hyphen (-) in the log file.
threadID	Identifier associated with the thread that generated the message. Use this field to track the progress of a single session or call when several requests are being processed simultaneously.
alarm	Optional field that is included only when the log message refers to an alarm notification. In this situation, the field describes the general category of the alarm. Alarm categories include: • Started • Quiesced • Shutdown • LicenseCheck • ConfigurationError • InitializationError • SotfwareException • InternalError • ResourceLimitation • CommunicationFailure • ProcessingFailure • InvalidArgument • UnexpectedEvent • NotificationDiscarded • Watchdog • Timeout
message	Text description of the logged occurrence.

VoiceXML Interpreter system logs

The VoiceXML Interpreter creates system logs that contains information about the active VoiceXML Interpreter process. The name format for the system log is:

*vxmlinterpreter_**creationdate**_[**index**].log*

where:

- ***creationdate*** is the date the system log file was created, formatted as local server time by default.
- ***index*** is an integer specifying the current incremented VoiceXML Interpreter system log file. This value is reset daily and incremented when the configured maximum system log file size is reached or when the VoiceXML Interpreter is restarted. Because the index value is a timestamp, indexes are ordered, but not necessarily consecutive.

For example, the following VoiceXML Interpreter system log file was generated by the vxmlinterpreter process on 2008-08-28. It has an index value of 5:

```
vxmlinterpreter_2008_08-28_005.log
```

This topic describes the following information about VoiceXML Interpreter system logs:

- Logging levels
- Logging defaults
- Changing the logging level
- Changing other logging defaults
- Log file format

Logging levels

The following table describes the logging levels for VoiceXML Interpreter system logs in decreasing severity and increasing verbosity order:

Logging level	Description
FATAL	Logs only critical errors.
ERROR	Logs all errors.
WARNING	Logs all errors and warnings.
INFO1	Logs information useful for first-level debugging and also logs some normal events.
INFO2 INFO3	Logs information useful for second-level debugging. Because this level generates abundant log information, you should set this level only at the request of Dialogic Technical Services and Support.
INFO4 INFO5	The most verbose option, which logs low-level detailed information. Because this level generates abundant log information, you should set this level only at the request of Dialogic Technical Services and Support.

Logging defaults

The default values for VoiceXML Interpreter system logs are:

- The logging level is set to INFO1. This is sufficient for routine logging on a production machine.
- The system log file is stored in the *vx/vxmlinterpreter/logs* directory.
- The maximum number of log files in the log directory is 50. If the log directory contains 50 log files, then the oldest log file is deleted when the 51st log file is added.
- The maximum size of a log file is 10 MB. When a log file reaches this size, a new log file is added.
- The date/time format for the log creation date and log entry date is in local server time.
- The index for the VoiceXML Interpreter system log file is reset each day. The index is incremented when a new log file is started because the maximum system log file size was reached or the 51st log file was added to the log directory.

For more information, see the *Dialogic® Vision™ VoiceXML Administration Manual*.

Changing the logging level

To change the VoiceXML Interpreter system logging level, follow these steps:

Step	Action
1	Access the Vision Console, as described in Accessing the Vision Console.
2	Click **Maintenance** on the Operations menu. The Maintenance page appears.
3	Select the desired log level in the Log level field associated with the VoiceXML Interpreter service, and click **Submit**.

Changing other logging defaults

The following table describes how to change the other logging defaults for the VoiceXML Interpreter system log:

To change the...	Modify the...
Log file location	LogDir setting in the *vxmlinterpreter.conf* file.
Maximum number of log files in the specified directory	SystemLogFileMaxNum setting in the *vxmlinterpreter.conf* file.
Maximum size of the log file	SystemLogFileMaxSize setting in the *vxmlinterpreter.conf* file.
Date/time format of the log file	SystemLogTime setting in the *vxmlinterpreter.conf* file.

For more information, see the *Dialogic® Vision™ VoiceXML Administration Manual*.

Log file format

The format of each VoiceXML Interpreter system log message is:

```
[timestamp] [severity] [category] [channel ID:thread ID] message
```

The following table describes the fields in a VoiceXML Interpreter system log message:

Field	Description
timestamp	Date and time of the log message, specified in local server time by default. Use the SystemLogTime setting in the *vxmlinterpreter.conf* file to change the time format to Greenwich Mean Time (GMT). This also changes the time format in the log file name. For information, see the *Dialogic® Vision™ VoiceXML Administration Manual*.
severity	Level of the log message. Valid values (in order of decreasing severity and increasing verbosity): • FATAL (a severe malfunction from which the VoiceXML Interpreter process cannot recover) • ERROR • WARNING • INFO1 (default) • INFO2 • INFO3 • INFO4 • INFO5 (highest/most verbose level of detail)

<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>category</td><td>General category to which the message refers:
<table>
<tr><th>Label</th><th>Description</th></tr>
<tr><td>Xml Parser</td><td>XML parser-related message.</td></tr>
<tr><td>VoiceXmlInterpreter</td><td>VoiceXML Interpreter flow/form interpretation algorithm related message.</td></tr>
<tr><td>I/O</td><td>Recognition/speech synthesis-related message.</td></tr>
<tr><td>Debug</td><td>Debug message.</td></tr>
<tr><td>Telephony</td><td>Telephony-related message.</td></tr>
<tr><td>Database</td><td>Database-related message.</td></tr>
<tr><td>JS</td><td>ECMA/JavaScript-related message.</td></tr>
<tr><td>Billing</td><td>Billing message.</td></tr>
<tr><td>Grammar</td><td>Grammar-related message.</td></tr>
</table></td></tr>
<tr><td>channel ID</td><td>ID of the VoiceXML Interpreter telephony channel that generated the message. Otherwise, the channel ID is replaced with a dash, such as for process-wide log messages.</td></tr>
<tr><td>thread ID</td><td>ID of the thread that generated the message. This can be used to track the progress of a single session in the log file when there are several requests being processed simultaneously.</td></tr>
<tr><td>message</td><td>Log message.</td></tr>
</table>

SSML Processor system logs

The SSML Processor creates system logs that contain information about the active SSML Processor process. The name format for the system log is:

*ssmlprocessor_**creationdate**_[**index**].log*

where:

- ***creationdate*** is the date the log file was created, formatted as local server time by default.
- ***index*** is an integer specifying the current incremented SSML Processor system log file. This value is reset daily and incremented when the configured maximum system log file size is reached or when the VoiceXML Interpreter is restarted. Because the index value is a timestamp, indexes are ordered, but not necessarily consecutive.

For example, the following SSML Processor system log file was generated by the vxmlinterpreter process on 2008-05-23. It has an index value of 5:

```
ssmlprocessor_2008_05-23_005.log
```

This topic describes the following information about SSML Processor system logs:

- Logging levels
- Logging defaults
- Changing the logging level
- Changing other logging defaults
- Log file format

Logging levels

The following table describes the logging levels for SSML Processor system logs in decreasing severity and increasing verbosity order:

Logging level	Description
FATAL	Logs only critical errors.
ERROR	Logs all errors.
WARNING	Logs all errors and warnings.
INFO1	Logs information useful for first-level debugging and also logs some normal events.
INFO2 INFO3	Logs information useful for second-level debugging. **Note:** Because this level generates abundant log information, you should set this level only at the request of Dialogic Technical Services and Support.
INFO4 INFO5	The most verbose option, which logs low-level detailed information. **Note:** Because this level generates abundant log information, you should set this level only at the request of Dialogic Technical Services and Support.

Logging defaults

The default values for SSML Processor system logs are:

- The logging level is set to INFO1. This is sufficient for routine logging on a production machine.
- The system log file is stored in the *vx/ssmlprocessor/logs* directory.
- The maximum number of log files in the log directory is 50. If the log directory contains 50 log files, then the oldest log file is deleted when the 51st log file is added.
- The maximum size of a log file is 10 MB. When a log file reaches this size, a new log file is added.

- The date/time format for the log creation date and log entry date is in local server time.
- The index for the SSML Processor system log file is reset each day. The index is incremented when a new log file is started because the maximum system log file size was reached or the 51st log file was added to the log directory.

For more information, see the *Dialogic® Vision™ VoiceXML Administration Manual.*

Changing the logging level

To change the SSML Processor system logging level, follow these steps:

Step	Action
1	Access the Vision Console, as described in Accessing the Vision Console.
2	Click **Maintenance** on the Operations menu. The Maintenance page appears.
3	Select the desired log level in the Log level field associated with the SSML Processor service, and click **Submit**.

Changing other logging defaults

The following table describes how to change the other logging defaults for the SSML Processor system log:

To change the...	Modify the...
Log file location	LogDir setting in the *ssmlprocessor.conf* file.
Maximum number of log files in the specified directory	SystemLogFileMaxNum setting in the *ssmlprocessor.conf* file.
Maximum size of the log file	SystemLogFileMaxSize setting in the *ssmlprocessor.conf* file.
Date/time format of the log file	SystemLogTime setting in the ssmlprocessor.conf file.

For more information, see the *Dialogic® Vision™ VoiceXML Administration Manual*.

Log file format

The format of each SSML Processor system log message is:

```
[timestamp] [severity] [category] [channel ID:thread ID] message
```

The following table describes the fields in an SSML Processor system log message:

<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>timestamp</td><td>Date and time of the log message, specified in local server time by default.

Use the SystemLogTime setting in the ssmlprocessor.conf file to change the time format to Greenwich Mean Time (GMT). This also changes the time format in the log file name. For information, see the Dialogic® Vision™ VoiceXML Administration Manual.</td></tr>
<tr><td>severity</td><td>Level of the log message.

Valid values (in order of decreasing severity and increasing verbosity):
FATAL (a severe malfunction from which the VoiceXML Interpreter process cannot recover)
ERROR
WARNING
INFO1 (default)
INFO2
INFO3
INFO4
INFO5 (highest/most verbose level of detail)</td></tr>
<tr><td>category</td><td>General category to which the message refers:<table><tr><th>Label</th><th>Description</th></tr><tr><td>TTS Engine</td><td>Messages related to TTS engines and their management.</td></tr><tr><td>Framework</td><td>Messages related to the general SSML Processor framework.</td></tr></table></td></tr>
<tr><td>channel ID</td><td>ID of the SSML Processor channel that generated the message, if the message came from an audio channel. Otherwise, the channel ID is replaced with a dash.</td></tr>
<tr><td>thread ID</td><td>ID of the thread that generated the message. This can be used to track the progress of a single SSML request in the log file when there are several requests being processed simultaneously.</td></tr>
<tr><td>message</td><td>Log message.</td></tr>
</table>

VoiceXML application logs

VoiceXML applications running on the VoiceXML Interpreter create log messages that VoiceXML application developers can use to debug phone calls.

By default, application logging is set up as follows:

- Application logging is enabled.
- A separate log file is created for each call.
- The application log files are stored in the *vx/vxmlinterpreter/logs/**subdir*** directory, where ***subdir*** is the log subdirectory specified in the application configuration. For information about configuring applications, see Managing VoiceXML applications.

The following table describes how to change the defaults for application logging:

To...	Take this action...	For more information, see...
Disable application logging for all applications	Set the VXMLAppLogsEnabled setting in the *vxmlinterpreter.conf* file to false.	*Dialogic® Vision™ VoiceXML Administration Manual*
Change the base location for all log files	Modify the VXMLAppLogBaseDir setting in the *vxmlinterpreter.conf* file.	*Dialogic® Vision™ VoiceXML Administration Manual*
Change the subdirectory location for a specific application's log file	Modify the setting of the Log subdirectory field on the VoiceXML application configuration page in the Vision™ Console.	Managing VoiceXML applications

Note: To facilitate debugging, you can also send VoiceXML application messages to the VoiceXML Interpreter system log file. To place application log messages in the system log file, set the value of SystemLogLevel in the *vxmlinterpreter.conf* file to 3 or higher.

Log file format

The format for the VoiceXML Interpreter application log file name is:

```
[VXMLAppLogBaseDir]/{subdir]/yyyy/mm/hh/[Call ID].log
```

where:

- ***subdir*** is the log subdirectory specified in the application configuration. For more information, see Managing VoiceXML applications.
- ***Call ID*** is the unique identifier for the current call.

The following table describes the fields in a VoiceXML Interpreter application log message:

Field	Description
timestamp	Date and time of the log message specified in UTC time as: `mm/dd/yy hh:mm:ss`

<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>category</td><td>Application category to which the message refers. The category types are:<table><tr><th>Label</th><th>Description</th></tr><tr><td>Recognition</td><td>Voice recognition message.</td></tr><tr><td>Events</td><td>VoiceXML event related message.</td></tr><tr><td>Variables</td><td>ECMA Script related message.</td></tr><tr><td>HTTP</td><td>HTTP message.</td></tr><tr><td>Log</td><td>VoiceXML <log> element message.</td></tr><tr><td>Flow</td><td>VoiceXML Interpreter flow/form Interpretation algorithm related message.</td></tr></table></td></tr>
<tr><td>severity</td><td>Level of severity associated with the log message.
Valid values:
ERROR
WARNING
INFO</td></tr>
<tr><td>message</td><td>Log message.</td></tr>
</table>

The following example shows a VoiceXML Interpreter application log message:

```
08/28/08 14:52:23 [Flow] [Info]
 Playing prompt http://localhost:9002/vxml/examples/helloworld/helloworld.wav
```

SSML application logs

Interspersed with the SSML Processor system log messages are higher-level application log messages that are used to debug speech applications. When the SSML Processor is used as the audio source for the Programmable Media Platform, these messages are sent using HTTP POST using the VoiceXML Interpreter process that generated the SSML request. The messages are inserted into the application log file when the request has been processed.

The format of these messages is:

[***timestamp***] [***category***] [***severity***] ***message***

The following table describes the fields in an SSML Processor application log message:

Field	Description
timestamp	Date and time of the log message specified in UTC time as: ***mm/dd/yy hh:mm:ss.mmm***

<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>category</td><td>Type of message:<table><tr><th>Label</th><th>Description</th></tr><tr><td>HTTP</td><td>Information on HTTP requests, responses, and caching.</td></tr><tr><td>Flow</td><td>Information on the general SSML Processor flow.</td></tr></table></td></tr>
<tr><td>severity</td><td>Level of severity associated with the message.
Valid values:
ERROR
WARNING
INFO</td></tr>
<tr><td>message</td><td>Log message.</td></tr>
</table>

Logging and performance

When the Programmable Media Platform actively uses large numbers of channels, a large amount of logging can be produced. Due to system limits, such as the input/output rate of the machine operating system when writing to the hard drive, this might cause performance bottlenecks.

Set the system log level appropriately for the VoiceXML Interpreter, Call Server, SSML Processor, and Media Resource Function logs. For example, use log level 1 or lower for a deployment environment. In general, if the Programmable Media Platform uses a higher number of concurrent channels, change the log levels to a lower level of verbosity.

Use SystemLogFileMaxSize and SystemLogFileMaxNum at their default settings to actively prune log files.

8. Authorization and Usage Indication interface

Overview of the Authorization and Usage Indication interface

The Authorization and Usage Indication interface is an XML-over-HTTP mechanism modeled on the ETSI TS 101 321 Open Settlement Protocol (OSP) for authorization and usage exchange.

An authorization request is made at the start of the session or call and before the session is started or the call is answered. The subsequent authorization response sent in the HTTP response can:

- Authorize the call
- Modify the DNIS of an authorized call
- Deny the call

Modifying the DNIS of an inbound call allows the platform operator to hide private number ranges by manually replacing the local number with the original assigned number. At the end of the session or call, the VoiceXML Interpreter dispatches a call detail record (CDR) that contains the following information about a call:

- Time the call started
- Time the call ended
- Length of the call
- Transferred call information

A usage response is sent in the HTTP response to a usage indication request. If there is a problem sending this response, the Programmable Media Platform writes call detail records (CDRs) to the local disk. This occurs when the billing server is temporarily unavailable. For information, see Call detail records.

Message format

The following example shows the general format for OSP messages:

```
<?xml  version="1.0"?>
 <Message  messageId="0"  random="1234">
  ...
 </Message>
```

The Message element contains one of the following child elements, called components. With the exception of UsageIndication, the components only appear once.

- AuthorizationRequest
- AuthorizationResponse
- UsageIndication
- UsageConfirmation

The following table describes the attributes of the Message element:

Attribute	Description
messageId (componentId)	Unique ID assigned to the message element and to its immediate descendants.
random	Random decimal value.

HTTP operations

The VoiceXML Interpreter uses the HTTP POST (application/x-www-form-urlencoded) operation with the field name osprequest. The field value contains the XML data contained in the AuthorizationRequest, AuthorizationResponse, UsageIndication, or UsageConfirmation messages. For more information, see Authorization exchange and Usage exchange.

Authorization exchange

The authorization exchange consists of the following types of messages:

- AuthorizationRequest
- AuthorizationResponse

AuthorizationRequest

The following example shows the structure of an AuthorizationRequest message:

```
<AuthorizationRequest  componentId="1">
    < Timestamp>
        2003-11-14T12:34:00Z
    </ Timestamp>
    < CallId>
        0001D747-318C9A10-B3DB0006-5B39B574
    </ CallId>
    < SourceInfo  type="e164">
        35312345678
    </ SourceInfo>
    < DestinationInfo  type="e164">
        35318765432
    </ DestinationInfo>
    <vision:CallType>
        INBOUND
    </vision:CallType>
   <Service/>
    < MaximumDestinations>
        0
    </ MaximumDestinations>
</AuthorizationRequest>
```

Note: The Programmable Media Platform strips extra white space from actual requests to improve efficiency.

The following table lists the XML elements used in an AuthorizationRequest message:

Element	Abbreviation	Description
Timestamp	ts	Time at which the authorization request was initiated. The format is in coordinated universal time (UTC): ***yyyy-mm-ddthh:mm:ss.sssz***. Millisecond accuracy is used.
CallId	ci	Globally unique identifier for the call that corresponds to the VoiceXML session variable session.connection.callid.
SourceInfo	si	ANI of the session or call. Valid values: • e164 - Default. Number format starts with the international prefix; for example, 3531234567. • uri - Number is a SIP URI.
DestinationInfo	di	DNIS of the session or call. Valid values: • e164 - Default. Number format starts with the international prefix; for example, 3531234567. • uri - Number is a SIP URI.
vision:CallType	vct	Indicates the type of call. Valid values: • INBOUND • OUTBOUND • TRANSFER
Service	N/A	Used according to Open Settlement Protocol (OSP) implementation, but not filled with values.
MaximumDestinations	N/A	Used according to OSP implementation, but not filled with values.

AuthorizationResponse

The following example shows the structure of an AuthorizationResponse message:

```
<AuthorizationResponse  componentId="1">
    < Timestamp>2003-11-14T12:34:00Z</ Timestamp>
    <Status>
        <Code>200</Code>
        <Description>success</Description>
    </Status>
    < TransactionId>000000</ TransactionId>
</AuthorizationResponse>
```

Note: The Programmable Media Platform strips extra white space from actual responses to improve efficiency.

The following table lists the XML elements used in an Authorization Response message:

Element	Abbreviation	Description
Timestamp	ts	Time at which the authorization response was initiated. The format is in coordinated universal time (UTC): Format: ***yyyy-mm-ddthh:mm:ss.sssz***. Millisecond accuracy is used.
Status	N/A	Valid values: • 200 - Call authorized • 403 - Call not authorized • 404 - DNIS unknown
vision:Masquerade	N/A	Optional. If present, the VoiceXML Interpreter replaces the DNIS with this number in the VoiceXML session variables and logs.
TransactionId	ti	Not used.
vision:FeaturePermission	N/A	Optional. Controls permission for a call to a particular VoiceXML Interpreter feature. Format: ***name value*** Valid values for ***name***: • asr • tts • transfer Valid values for ***value***: • true - Default. Allows the call to use the named feature. • false - Disallows the call from using the named feature.

Usage exchange

The usage exchange consists of the following types of messages:

- UsageIndication request
- UsageConfirmation

UsageIndication request message

The following example shows the UsageIndication request message format:

```
<UsageIndication  componentId="1">
    <Timestamp>
        2003-11-14T12:34:45Z
    </Timestamp>
    <Role>
        source
    </Role>
    <TransactionId>
        000000
    </TransactionId>
    <CallId>
        0001D747-318C9A10-B3DB0006-5B39B574
    </CallId>
    <SourceInfo  type="e164">
        35312345678
    </SourceInfo>
    <DestinationInfo  type="e164">
        35318765432
    </DestinationInfo>
    <vision:CallType>
        INBOUND
    </vision:CallType>
    < UsageDetail>
        <Service/>
        <Amount>
            45
        </Amount>
        <Increment>
           1.000000
        </Increment>
        <Unit>
            s
        </Unit>
        < StartTime>
            2003-11-14T12:34:00.000Z
        </ StartTime>
        < EndTime>
            2003-11-14T12:34:45.000Z
        </ EndTime>
        < TerminationCause>
            < TCCode>
                1016
            </ TCCode>
            <Description>
                Normal call clearing
            </Description>
        </ TerminationCause>
    </ UsageDetail>
</UsageIndication>
```

Note: Blind transfers do not appear as additional UsageIndication elements.

The following table lists the XML elements used in the UsageIndication message:

Element	Description
Timestamp	Time at which the authorization request was initiated. The format is in coordinated universal time (UTC): ***yyyy-mm-ddthh:mm:ss.sssz***. Millisecond accuracy is used.

Element	Description
Role	Required for a usage exchange message under the Open Settlement Protocol (OSP), but has no value.
TransactionId	Required for a usage exchange message under the Open Settlement Protocol (OSP), but has no value.
CallId	Globally unique identifier for the call that corresponds to the VoiceXML session variable session.connection.callid.
SourceInfo	ANI of the session (call). Valid values: • e164 - Default. Number format starts with the international prefix; for example, 3531234567. • uri - Number is a SIP URI.
DestinationInfo	DNIS of the session (call). Valid values: • e164 - Default. Number format starts with the international prefix; for example, 3531234567. • uri - Number is a SIP URI.
vision:CallType	Type of call. Valid values: • INBOUND • OUTBOUND • TRANSFER
vision:ParentCallID	For transferred calls, a child element of the UsageIndication message (component) that refers to the CallID of the original call. The original call can be inbound or outbound. The first UsageIndication message refers to the original inbound or outbound call. Subsequent UsageIndication messages, if present, refer to transferred calls and contain the <vision:ParentCallID> element. **Note:** An additional UsageIndication is not created for a blind transfer.
vision:ServiceNumber	A Programmable Media Platform service number for outbound calls.
UsageDetail	A block that contains details about a call. If an external system does not require this information, use the BillingOmitUsageDetail configuration setting in the *vxmlinterpreter.conf* file to omit it. For more information, see the *Dialogic® Vision™ VoiceXML Administration Manual.*

Element	Description
Service	Required for a usage exchange message under the Open Settlement Protocol (OSP), but has no value.
Amount	Duration of the call in seconds, rounded to nearest increment, as specified by the <Increment> element.
Increment	Increment in seconds for which values are output for the <StartTime>, <EndTime>, and <Amount> elements in the UsageDetail block. The value is always set to 1.000000. For example, if a call started at 2003-11-14T12:34:00.550Z, the time would be output in the <StartTime> element as: `<StartTime>2003-11-14T12:34:01.000Z</StartTime>`
Unit	Time designator associated with the Amount and Increment elements in the UsageDetail block. The value is always set to s (seconds).
StartTime	Time the call started, specified in the same format as the Timestamp element.
EndTime	Time the call ended, specified in the same format as the Timestamp element.
TerminationCause	Termination reason for a call, which includes a code in the <TCCode> element and plain text in the <Description> element.
TCCode	Termination code for a call. Valid values: • 1016 - Normal call clearing • 0017 - Busy • 0018 - No answer • 0038 - Network out of order • 0041 - Telephony error (system error)

UsageConfirmation message

The following example shows the UsageConfirmation message format:

```
<UsageConfirmation  componentId="1">
    <Timestamp>
        2003-11-14T12:34:45Z
    </Timestamp>
    <Status>
        <Code>
            201
        </Code>
        <Description>
             cdr created
        </Description>
```

```
        </Status>
</UsageConfirmation>
```

The following table lists the XML elements used in the UsageConfirmation message:

Element	Description
Timestamp	Time the usage indication message request was initiated. Format: ***yyyy-mm-ddthh:mm:ss.sssz***. Millisecond accuracy is used.
Status Code	Valid values: 201 - Success 500 - Error occurred

If there is a problem sending a UsageConfirmation message, the Programmable Media Platform writes call detail records (CDRs) to the local disk. This occurs when the billing server is temporarily unavailable. For information, see Call detail records.

Call detail records

By default, the Programmable Media Platform writes call detail records (CDRs) to disk when any of the following actions occur:

- CCXML session ends
- Endpoint hangs up the phone
- The billing server is temporarily unavailable.

You can direct the Programmable Media Platform to always record CDRs to disk, even when the Usage Indication requests are successful. To do this, set the BillingWriteCDRToDisk setting in the *vxmlconfig.conf* file to true.

The Programmable Media Platform records CDRs into a single text file in a condensed format. The CDR has a directory structure and name that uses the following format:

YYYY*/*DD*/*MM*/<*hour*>.*cdr

The CDR file rolls over on the hour, every hour using UTC time. For example, the CDR file called *2009/08/06/05.cdr* is the file recorded at 5 am on the 8th of June 2009.

When a Usage Indication request is not fulfilled, the OSP XML CDR will be converted to the CDR file entry format and appended to the single CDR file. If the CDR file does not exist, it is created.

You can use a tool to parse these CDR files and attempt an upload of the CDRs to the Billing server at a later time when it becomes available again.

To view CDR files, use the **CDR files** option on the Monitoring menu. For more information, see CDR files.

CDR entry format

The Programmable Media Platform uses the following format for CDR entries, with multiple name/value pairs separated by commas:

```
<name>=<value>,<name>=<value>,...;
```

A CDR entry omits fields that are not present in order to aid with parsing and disk-space efficiency.

If a CDR entry has been made in the file as a result of a failed Usage Indication request, the @ character is used as a suffix. This differentiates this type of CDR entry from CDRs entered because the Vision Server was configured to record all CDRs.

The following example shows a complete CDR entry:

```
ci=B28584CD 3B5011D9 80990007 EB592A8A,ts=2004-11-22T00:03:12Z,tc=1016,
dn=Normal call clearing,si=0035312091912,se='e164',di=170363161,dt='e164',
vct=INBOUND,vcm=VIDEO,vsn=170363161,it=1.000000,rl=source,at=15,ti=000000,ut=s,
st=2004-11-22T00:02:56.353Z,et=2004-11-22T00:03:11.382Z;\n
```

The following table illustrates the mapping between CDR abbreviations and Open Settlement Protocol (OSP) XML elements:

Abbreviation	OSP XML element	Description
at	Amount	Call duration, when this value is multiplied by the value of the Increment element.
ci	CallId	Call identifier.
di	DestinationInfo	Destination information or DNIS in one of the following formats: • Full ITU-T Recommendation E.164 telephone number (11 numeric digits) • URL (12 digits) For example: `di=1290`
dn	Description	Call description.
dt	DestinationType	Destination type. Defaults to e64 or to a uri.
et	EndTime	Service end time, in GMT format. For example: `et=2007-11-08T08:00:58.345Z`
it	Increment	Increment of service measurement.
rl	Role	Role of the system generating the message. Valid values: • source • destination • other
se	SourceInfoType	Source type. Defaults to e164 or can be a URI. For example: `se=e164`
si	SourceInfo	Source information or ANI.

Abbreviation	OSP XML element	Description
st	StartTime	Service start time, in GMT format. For example: `st=2007-11-08T08:00:51.355Z`
tc	TerminationCauseCode	Termination code. Valid values: • 1016: Normal call clearing • 0017: Busy • 0018: No answer • 0038: Network out of order • 0041: Telephony error (system error)
ti	TransactionId	Transaction identifier.
ts	Timestamp	Time at which the component was generated, in GMT format.
ut	Unit	Units in which usage is recorded. Valid values: • s: seconds • pkt: packets (data grams) • b: bytes • p: fax pages • call:calls
vcm	vision:CallMode	Call mode. Valid values: • Voice • Video
vct	vision:CallType	Call type. Valid values: • Inbound • Outbound • transfer

Abbreviation	OSP XML element	Description
vpc	vision:ParentCallID	For transferred calls, a child element of the UsageIndication message (component) that refers to the CallID of the original call. The original call can be inbound or outbound. The first UsageIndication message refers to the original inbound or outbound call. Subsequent UsageIndication messages, if present, refer to transferred calls and contain the <vision:ParentCallID> element. **Note:** An additional UsageIndication is not created for a blind transfer.

9. SIP interface

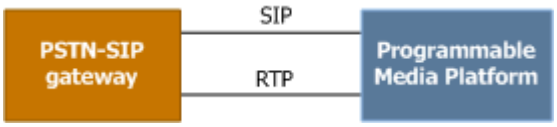
Overview of the SIP interface

The Programmable Media Platform integrates with SIP in two ways:

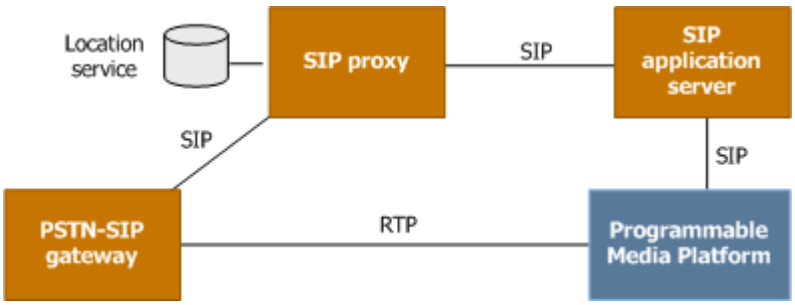
- Directly terminates IP calls with SIP signaling.
- Acts as a slave to application servers that establish RTP and VoiceXML sessions.

The following illustration shows how the Programmable Media Platform terminates IP calls:

- Directly terminates IP calls with SIP signaling.
- Acts as a slave to application servers that establish RTP and VoiceXML sessions.



The following illustration shows how the Programmable Media Platform integrates into a typical SIP network involving application servers, proxy servers, location servers, and servers:



The Programmable Media Platform SIP interface is described in more detail in the following topics:

- Inbound calls
- ISUP to SIP cause values
- SIP to ISUP cause values
- VoiceXML exit information
- BYE reason header
- Call transfer

Inbound calls

An inbound VoiceXML session is initiated through a SIP INVITE. The SDP Offer/Answer model (RFC 3264) is used for media negotiation. The INVITE usually contains the offer and the 200 OK response contains the answer SDP.

The following table lists the possible responses to the SIP INVITE:

Response	Description
200 OK	Success.
480 Temporarily Unavailable	Call not authorized.
503 Service Unavailable	No channels available to take call.

By default, the Programmable Media Platform handles incoming DTMF content in the following manner:

- If RFC 2833 is negotiated, then the server obtains the DTMF tone and duration through an RFC 2833 DTMF event.
- If RFC 2833 is not negotiated, then the server obtains the DTMF tone and duration from the body of the SIP INFO message.

If the joined connection is a SIP connection, then audio transcoding automatically occurs between the two endpoints, when required.

Variable mappings

The following table lists the SIP INVITE components and the corresponding VoiceXML session variables:

INVITE component	VoiceXML session variable
INVITE Protocol Name	session.connection.protocol.name (evaluates to sip)
INVITE Protocol Version	session.connection.protocol.version (evaluates to 2.0)
Request-URI	session.connection.protocol.sip.requesturi
Request-URI aai parameter	session.connection.aai (Also available through session.connection.protocol.sip.requesturi.aai)
Request-URI ccxml parameter	Stored as sub-properties of the session.connection.ccxml object (Also available through session.connection.protocol.sip.requesturi.ccxml)

INVITE component	VoiceXML session variable
Request-URI voicexml parameter	session.connection.protocol.sip.requesturi.voicexml (Also available through session.connection.initialuri) **Note:** If a query string is included in the parameter value, URL encode the special parameters. For example, replace the ***?*** and **=** characters with ***%3F*** and ***%3D*** respectively.
Other parameters of the SIP Request-URI	Stored as sub-properties of the session.connection.protocol.sip.requesturi object
To header	session.connection.local.uri
From header	session.connection.remote.uri
Call-ID header	session.connection.callid
All headers of INVITE	Stored as sub-properties of thesession.connection.protocol.sip.headers object

When set to true (the default), the media resource object (MIOSIP) configuration setting com.vision.miosip.dialog.FromToUseEntireSIPURI enables backward compatibility for VoiceXML applications that use the session.connection.remote.uri and session.connection.local.uri variables.

When MIOSIP receives a SIP BYE message, the Reason header (if present) is mapped to the _message variable associated with the VoiceXML connection.disconnect.hangup event.

Specifying the initial VoiceXML URI to invoke

To specify the initial VoiceXML URI to invoke, use the NETANN procedure. Include the VoiceXML URI to invoke as the ***voicexml*** parameter of the SIP Request-URI. If you omit the ***voicexml*** parameter, the Programmable Media Platform uses its application configuration system to map the user part of the To: field to an initial VoiceXML page.

For example:

```
sip:dialog@mediaserver.example.net; \
voicexml=http://vxmlserver.example.net/cgi-bin/script.vxml;aai=information
```

Session data update

You can obtain additional session protocol information after receiving a 200 OK response and answering a call; for example, to obtain the name of a videophone vendor after the underlying telephony connection is established.

To send additional information, use the Vision-OrigProtocolInfo header as part of the ACK to the 200 OK.

The syntax for the Vision-OrigProtocolInfo header is:

```
Vision-OrigProtocolInfo: name=sip,version=2,mode=video
```

This data maps to the session.connection.protocol.sip.headers["Vision-Origprotocolinfo"] session variable.

CCXML parameters

Since a CCXML application can initiate a session with the Programmable Media Platform, CCXML defines the following session variables within VoiceXML:

- session.connection.ccxml.sessionid
- session.connection.ccxml.dialogid
- session.connection.ccxml.connectionid
- session.connection.ccxml.conferenceid
- session.connection.ccxml.namelist
- session.connection.ccxml.values

These variables originate from parameters of the Request-URI without the session.connection prefix:

```
sip:dialog@10.0.0.1;ccxml.sessionid=1234;ccxml.dialogid=1234;ccxml.values.var1=value1
```

The session.connection.ccxml.namelist is not explicitly passed in the Request-URI. The Programmable Media Platform reconstructs the namelist from the ccxml.values parameters.

The session.connection.ccxml parameters are also available through the session.connection.protocol.sip.requesturi["ccxml"] session variable.

Proxy server

You can configure the VoiceXML Interpreter to use the proxy server specified in the com.vision.miosip.rvsip.outboundProxyHost configuration setting of the *vxmlinterpreter.conf* file. For more information, see the *Dialogic® Vision™ VoiceXML Administration Manual*.

Location server

The VoiceXML Interpreter can register with a location server, which is often co-located with the SIP proxy. To register the VoiceXML Interpreter with a location server, use the REGISTER method in conjunction with the configuration settings described in the following table:

Setting	Description
com.vision.miosip.rvsip.locationServerAddress	URI of the location server.
com.vision.miosip.rvsip.userAgentAoR	SIP address of record for the Programmable Media Platform
com.vision.miosip.rvsip.userAgentContact	Contact address.

Note: When the VoiceXML Interpreter is quiesced, it de-registers itself from the location server to allow for a graceful shutdown.

Load balancing and failover

Through its support of RFC 3263 behavior, the Programmable Media Platform provides resource discovery, load balancing, and transparent failover. The Programmable Media Platform uses the following configuration settings in the *vxmlinterpreter.conf* file to implement load balancing and failover behaviors:

- com.vision.miosip.dialog.RemoteHosts

- com.vision.miosip.rvsip.outboundProxyHost
- com.vision.miosip.rvsip.locationServerAddress

You can specify a comma-delimited list of IP addresses in the com.vision.miosip.dialog.RemoteHosts setting. The VoiceXML Interpreter performs a random robin algorithm on the list. If no response is received from an INVITE, the VoiceXML Interpreter transparently tries another address in the list. For more information, see the Dialogic® *Vision™ VoiceXML Administration Manual*.

For unreliable transport protocols such as UDP, the retransmission timer for an INVITE occurs with an exponential backoff of 2^n * T1 and a maximum upper bound of 64 * T1. Even with a recommended reduced setting for T1 (500ms) for high QoS networks, the initial INVITE takes 32 seconds to time out if the target server is not operational. TCP is preferable because the transport layer detects the error sooner and allows for faster failover. To use persistent TCP connections, set the com.vision.miosip.rvsip.ePersistencyLevel configuration setting appropriately.

ISUP to SIP cause values

The following table shows the default ISUP cause code to SIP response. This mapping follows RFC 3398, ISUP to SIP Mapping.

Note: If a cause value other than those listed below is received, the default response '500 Server internal error' should be used.

ISUP cause value	SIP response
Normal event	
1 Unallocated number	404 Not found
2 No route to network	404 Not found
3 No route to destination	404 Not found
16 Normal call clearing	(Typically results in a BYE or CANCEL)
17 User busy	486 Busy here
18 No user responding	408 Request timeout
19 No answer from user	480 Temporarily unavailable
20 Subscriber absent	480 Temporarily unavailable
21 Call rejected	403 Forbidden
22 Number changed (without diagnostic)	410 Gone
22 Number changed (with diagnostic)	301 Moved permanently
23 Redirection to new destination	410 Gone

ISUP cause value	SIP response
26 Non-selected user clearing	404 Not found
27 Destination out of order	502 Bad gateway
28 Address incomplete	484 Address incomplete
29 Facility rejected	510 Not implemented
31 Normal unspecified	480 Temporarily unavailable or Resource unavailable
34 No circuit available	503 Service unavailable
38 Network out of order	503 Service unavailable
41 Temporary failure	503 Service unavailable
42 Switching equipment congestion	503 Service unavailable
47 Resource unavailable	503 Service unavailable
55 Incoming calls barred within CUG	403 Forbidden
57 Bearer capability not authorized	403 Forbidden
58 Bearer capability not presently available	503 Service unavailable
65 Bearer capability not implemented	488 Not acceptable here
70 Only restricted digital bearer capability available (national use)	488 Not acceptable here
79 Service or option not implemented	501 Not implemented
Invalid message	
87 User not member of CUG	403 Forbidden
88 Incompatible destination	503 Service unavailable
102 Call setup time-out failure	504 Gateway timeout
111 Protocol error, unspecified	500 Server internal error
127 Interworking, unspecified	500 Server internal error

ISUP cause value	SIP response
Other	
	500 Server internal error (default)

SIP to ISUP cause values

The following table shows the default SIP response to ISUP cause code. This mapping follows RFC 3398, ISUP to SIP Mapping.

SIP response	ISUP cause value
400 Bad request	41 Temporary failure
401 Unauthorized	21 Call rejected
402 Payment required	21 Call rejected
403 Forbidden	21 Call rejected
404 Not found	1 Unallocated number
405 Method not allowed	63 Service or option unavailable
406 Not acceptable	79 Service/option not implemented
407 Proxy authentication required	21 Call rejected
408 Request timeout	102 Recovery on timer expiry
410 Gone	22 Number changed (without diagnostic)
413 Request Entity too long	127 Interworking
414 Request-URI too long	127 Interworking
415 Unsupported media type	79 Service/option not implemented
416 Unsupported URI Scheme	127 Interworking
420 Bad extension	127 Interworking
421 Extension Required	127 Interworking
423 Interval Too Brief	127 Interworking

SIP response	ISUP cause value
480 Temporarily unavailable	18 No user responding
481 Call/Transaction Does not Exist	41 Temporary Failure
482 Loop Detected	25 Exchange - routing error
483 Too many hops	25 Exchange - routing error
484 Address incomplete	28 Invalid Number Format
485 Ambiguous	1 Unallocated number
486 Busy here	17 User busy
487 Request Terminated	--- (no mapping)
488 Not Acceptable here	--- by Warning header
500 Server internal error	41 Temporary failure
501 Not implemented	79 Not implemented, unspecified
502 Bad gateway	38 Network out of order
503 Service unavailable	41 Temporary failure
504 Server time-out	102 Recovery on timer expiry
504 Version Not Supported	127 Interworking
513 Message Too Large	127 Interworking
600 Busy everywhere	17 User busy
603 Decline	21 Call rejected
604 Does not exist anywhere	1 Unallocated number
606 Not acceptable	--- by Warning header

VoiceXML exit information

Variables from the VoiceXML <exit> element are returned in the SIP BYE body. To allow the application server to differentiate between a BYE resulting from a <disconnect> and one resulting from an <exit>, an application can use either parameter _reason=disconnect or

_reason=exit. This is one way that a CCXML application can communicate with the Programmable Media Platform.

Note: An application can use either the expr attribute or the namelist attribute with the <exit> element, but not both.

There are two formats in which data can be returned to application server:

- Encoded according to SIP-VXML
- Formatted as XML

Returned data encoded as SIP-VXML

If the ExitDataXML configuration setting in the *vxmlinterpreter.conf* file is set to false, then returned data is encoded as specified in SIP-VXML.

Namelist variables are first converted to a string and encoded in the message body using the application/x-www-form-urlencoded format content type. The behavior resulting from specifying an ECMAScript object or a recording variable in the namelist is not defined.

The following examples show how data is returned encoded as SIP-VXML. For more information about SIP-VXML encoding, see SIP Interface to VoiceXML Media Services, Burke *et al*., IETF Internet-Draft, draft-burke-vxml-2, November 2006.

Example 1

This example shows how the value of the expr attribute of VoiceXML <exit> is encoded and placed in the SIP BYE message:

VoiceXML:

```
<exit expr="'OK'"/>
```

Data in SIP BYE message:

```
_exit=OK&_reason=exit
```

Example 2

This example shows how the value of the namelist attribute of VoiceXML <exit> is encoded and placed in the SIP BYE message:

VoiceXML:

```
<exit namelist=" myvar anothervar"/>
```

Data in SIP BYE message:

```
myvar=1&anothervar=2&__reason=exit
```

Returning data in XML format

If the ExitDataXML configuration setting in the *vx/vxmlinterpreter/conf* file is set to true, then data is returned in XML form. In this case, the MIME type for the body data is application/vxml-data+xml.

The following examples show how data is returned in XML format.

Example 1

In this example, the value attribute of <return> is set to the ECMA string value of the expr attribute of VoiceXML <exit>:

VoiceXML:

```
<exit expr="’OK’"/>
```

Data in SIP BYE message:

```
<vxml-data version="1.0" xmlns="http://www.dialogic.com/2004/vxml-data">
    <return value="OK"/>
</vxml-data>
```

Example 2

In this example, the format of the namelist content follows the XML serialization mechanism specified in Semantic Interpretation for Speech Recognition (SISR), with the top-level ECMAScript variable names serialized. The namelist content comes from the namelist attribute for <exit>.

VoiceXML:

```
<exit namelist="myvar myobject anothervar"/>
```

Data in SIP BYE message:

```
<vxml-data version="1.0" xmlns="http://www.dialogic.com/2004/vxml-data">
    <namelist>
        <myvar>1</myvar>
        <myobject>
            <x>1</x>
            <y>2</y>
        </myobject>
        <anothervar>2</anothervar>
     </namelist>
</vxml-data>
```

BYE reason header

When the Programmable Media Platform receives a BYE message, if the Reason header is present, it is mapped to the _message variable associated with the VoiceXML connection.disconnect.hangup event. This feature is used for communicating the termination reason, such as fax detection.

Call transfer

The Programmable Media Platform provides the following types of call transfers:

- Bridge
- Blind
- Consultation

Note: In this topic, the A leg refers to the initial call made to or from the Programmable Media Platform.

Bridge transfer

A bridge transfer occurs when the Programmable Media Platform initiates a transfer between two parties, and the caller returns to the Programmable Media Platform after the transfer ends (when the caller disconnects from the third party). With a bridge transfer:

- The Programmable Media Platform is aware of the outcome of the transfer.
- The original caller is not disconnected in the event of a connection error.

A bridge transfer appears as a new SIP INVITE from the Programmable Media Platform. The audio mixing occurs in the Media Resource Function component of the VoiceXML Interpreter. This component also performs DTMF hotword detection on the A leg.

The dest attribute has different meanings, depending on whether the call is being transferred to a SIP target or a PSTN target:

For this transfer target...	The dest attribute specifies the...
SIP	Destination SIP URI of the transfer target. Place this SIP call by using the com.vision.miosip.rvsip.outboundProxyHost configuration setting.
PSTN	User part of the To SIP URI. The remainder of the URI is constructed using the URI from the com.vision.miosip.dialog.TelHosts configuration setting. The dest attribute is specified using the tel: URI syntax and can include the custom parameters connecttimeout and maxtime. A tel: URI is converted to a SIP URI, with the user=phone parameter indicating that the call must be placed over the PSTN network.

The following example shows a tel: URI, along with its custom parameters:

```
tel:12345678;maxtime=60s&connecttimeout=10s.
```

The following table describes the custom parameters:

Custom parameter	Description
connecttimeout	Amount of time to wait for a final SIP response to the INVITE. When this time limit is reached, a CANCEL is issued and the transfer aborted. The timer for the connecttimeout starts when the 180 Ringing response is received.
maxtime	Maximum duration allowed for a call. When this time limit is reached, a BYE is issued to the outbound leg. The SIP INVITE contains the custom header Vision-ParentCallID, which equates to the Call-ID of the A leg.

While the outbound call is being attempted, the A leg hears the audio specified by the com.vision.miosip.defaultTransferAudio configuration setting.

The following table describes the mapping of SIP responses from the INVITE to the value of either the transfer form item variable or the VoiceXML event:

SIP response	Transfer form item variable / VoiceXML event
404 Not Found	error.connection.baddestination
408 Request Timeout	noanswer
486 Busy Here	busy
500 Server Internal Error	unknown

SIP response	Transfer form item variable / VoiceXML event
503 Service Unavailable	noresource
603 Decline	noanswer

The following table describes how the transfer form item variable is set when the outbound leg of a call is terminated:

Action	Transfer form item variable setting
If the called party hangs up (issues a BYE), the outbound call terminates.	far_end_disconnect
VoiceXML Interpreter terminates the outbound leg because the maximum call duration expired or because a hotword was detected.	near_end_disconnect

Blind transfer

A blind transfer occurs when the Programmable Media Platform initiates a transfer between two parties and then detaches from the call *before* the transfer takes place. With a blind transfer:

- The Programmable Media Platform is not aware of the outcome of the transfer.
- The original caller is disconnected if there is a transfer error.

A blind transfer uses the SIP REFER method (RFC 3515). After a successful response to the REFER message, the Programmable Media Platform generates a connection.disconnect.transfer and issues a BYE to drop the call.

The following table describes the attributes of the transfer element:

Attribute	Description
dest	Sets the user part of the SIP URI specified by the Refer-To header in the REFER message.
aai	Sets the aai part of the SIP URI specified by the Refer-To header in the REFER message.

The following example shows the use of a REFER message:

```
REFER sip:4708@10.3.1.52:5060 SIP/2.0
From: <sip:2078@dialogic.com>
To: <sip:10.3.1.52>;tag=ds-42d6-bb2c
Contact: sip:2078@10.3.2.89:5060
Call-ID: 12a1231bc323429@dialogic.com
CSeq: 1 REFER
Content-Length: 0
Refer-To: sip:1234568@voxbuilder.com;connecttimeout=2s;maxtime=60s
Via: SIP/2.0/UDP 10.0.0.99:5060
```

Valid responses to the REFER message are:

- 202 Accepted

- 500 Server Internal Error if an error occurs

Consultation transfer

A consultation transfer occurs when the Programmable Media Platform initiates a transfer between two parties and then detaches from the call after the transfer takes place. With a consultation transfer:

- The Programmable Media Platform is aware of the outcome of the transfer.
- The original caller is not disconnected in the event of a connection error.

A consultation transfer is also called a supervised transfer.

The following table describes the consultation transfer process:

Task	Description
1	The Programmable Media Platform creates a new call leg by issuing an INVITE message to the outbound server.
2	If the server accepts the call, the Programmable Media Platform issues a REFER message on the new call leg. The Refer-To header contains a Replaces field that holds the Call ID value for the new leg. This causes the new call leg to be replaced by the initial call leg on the remote UAC.
3	Control returns to the VoiceXML context.
4	A <connection.disconnect.transfer> event is issued.

10. Fine tuning the configuration

Overview of fine tuning the configuration

You should use the Vision Console to configure the Programmable Media Platform. In some circumstances, you may need to manually fine tune some of the server's configuration files. Before doing so, be sure to review the information in Avoiding conflicts with the Dialogic Vision Console.

Note: The manual method of updating configuration files is intended for advanced users and should be used in consultation with Dialogic Technical Services and Support. Inappropriate configuration may prevent the server from functioning normally.

The following topics provide more information on fine tuning the server configuration:

- Avoiding conflicts with the Dialogic Vision Console
- Fine tuning the H.100 clocking configuration

For information about using the Vision Console to configure the Programmable Media Platform, see Overview of configuring the Programmable Media Platform.

Avoiding conflicts with the Dialogic Vision Console

Under normal operation, only the Vision Console service can modify the server's configuration files. Any change you make manually to a configuration file while a user is active at the console may be overwritten by the Vision Console service.

Note: The manual method of updating configuration files is intended for advanced users and should be used in consultation with Dialogic Technical Services and Support. Inappropriate configuration may prevent the server from functioning normally.

To ensure that a manual change to a configuration file is preserved, follow these steps:

Step	Action
1	Make sure that all users are logged out of the Vision Console.
2	Manually change the required configuration file(s).
3	Login to the Vision Console.
4	Restart all services to activate the changes.

Fine tuning the H.100 clocking configuration

Note: This topic applies to all Programmable Media Platform models except IP-only audio and IP-only video.

The Programmable Media Platform includes an H.100 clock manager that synchronizes the server's boards. By default, the H.100 clock manager starts automatically when the Call Server starts up, if the Programmable Media Platform has more than one media board.

Default H.100 clocking configuration

By default, H.100 clocking is configured for standalone operation (Clocking.HBus.ClockMode=STANDALONE), as specified in the *oamsys.cfg* file.

Clocking configuration for ISUP models

For ISUP models, the signaling timeslot is split on different trunks and boards for reliability. There is one signaling trunk defined on each media board. The following table describes the default clocking configuration for an ISUP system with one or two media boards:

Clock component	Default configuration
Primary master	Set on media board 1 and drives A_CLOCK. Uses the most reliable signaling trunk on the given media board as the first timing reference. Falls back to NETREF.
Secondary master	Set on media board 2 and drives B_CLOCK. Uses the primary master as the first timing reference. Falls back to the most reliable trunk on the given media board, other than the one driving the primary master. **Note:** For servers with one media board, there is no secondary master.
Signaling board	Slave on A_CLOCK. Falls back to B_CLOCK. **Note:** For servers with dedicated signaling links, the signaling board is not connected on the H.100 bus.

Clocking configuration for ISDN models

For ISDN models, all trunks carry a signaling link (D channel) for synchronization. The following table describes the default clocking configuration for an ISDN system with two media boards:

Clock component	Default configuration
Primary master	Set on media board 1 and drives A_CLOCK. Uses the most reliable signaling trunk on the given media board as the first timing reference. Falls back to NETREF.
Secondary master	Set on media board 2 and drives B_CLOCK. Uses the primary master as the first timing reference. Falls back to the most reliable trunk on the given media board, other than the one driving the primary master. **Note:** For servers with one media board, there is no secondary master.

H.100 clock manager configuration file

The H.100 clock manager uses the *priorities.xml* file to configure log settings, wait timeout intervals, timing references, and the board index of the clock source. This file resides in the *vx/clockmgr* directory.

The following example shows the default *priorities.xml* file for the ISUP models:

```
<?xml version="1.0" encoding="utf-8" ?>
<clockmanager>
    <log level="INFO" basefilename="clockmgr" outputdir="logs" filesize="1MB"
        maxfiles="5" />
    <txboard boardindex="1" />
    <clocksources>
        <source boardindex="0" trunk="1" priority="0" />
        <source boardindex="1" trunk="1" priority="0" />
    </clocksources>
</clockmanager>
```

The following table describes the elements and attributes in the *priorities.xml* file:

Element	Description
<clockmanager>	Root element of the *priorities.xml* configuration file.
<clocksources>	Root element for all timing references. This element is a child of the <clockmanager> element. All elements besides <clockmanager> are children of the <clocksources> element.

<table>
<tr><th>Element</th><th>Description</th></tr>
<tr><td><log></td><td>Log settings for the H.100 clock manager. All attributes are optional.
<table>
<tr><th>Attribute</th><th>Description</th></tr>
<tr><td>level</td><td>Log level for the H.100 clock manager log.
Valid values:
CRITICAL
ERROR
WARNING
INFO
Default: ERROR</td></tr>
<tr><td>basefilename</td><td>Base file name of the log output files.
Default: clkmgr</td></tr>
<tr><td>outputdir</td><td>Output directory for the log files.
Default: vx/clockmgr/logs</td></tr>
<tr><td>filesize</td><td>File size of each log file. Include a unit identifier (B, KB, or MB) with the value. If you do not include a unit identifier, the Media Platform uses MB.
Valid values: 1B - nMB
Default: 1MB
Valid values: 1B - nMB
Default: 1MB</td></tr>
<tr><td>maxfiles</td><td>Maximum number of files in the H.100 clock manager log. When all files are full, the logging subsystem overwrites the contents of the first file.
Valid values: 1 - n
Default: 5</td></tr>
</table></td></tr>
</table>

<table>
<tr><th>Element</th><th>Description</th></tr>
<tr><td><frequency></td><td>Wait timeout for events. If defined, the clocking service executes in polling mode. If not defined, the clocking service executes in asynchronous mode (recommended).
<table>
<tr><th>Attribute</th><th>Description</th></tr>
<tr><td>value</td><td>Time period, in milliseconds, for which the main loop waits for events before sending the timeout event.
Valid values:-1 - n
Default: -1 (CTA_WAIT_FOREVER)
Required: No.</td></tr>
</table></td></tr>
<tr><td><source></td><td>Timing reference for H.100 bus clocking. All attributes are required.
<table>
<tr><th>Attribute</th><th>Description</th></tr>
<tr><td>boardindex</td><td>OAM board index referenced by this clock source.
Valid values: 0 - n
Default: None.</td></tr>
<tr><td>trunk</td><td>Trunk number for the clock source.
Valid values: 0 - n, where n is the total number of trunks supported by the board type. A value of 0 designates the board's internal oscillator (OSC) as the clock source.
Default: None.</td></tr>
<tr><td>priority</td><td>Reliability of the trunk specified in the trunk attribute. Trunks with equivalent reliability can be given identical priority numbers.
Valid values: 0 (best) - 99 (worst)
Default: None.</td></tr>
</table></td></tr>
</table>

<table>
<tr><th>Element</th><th>Description</th></tr>
<tr><td><txboard></td><td>(ISUP models only) Defines the signaling board in the system. This element only applies when signaling is embedded in the voice trunks.
<table>
<tr><th>Attribute</th><th>Description</th></tr>
<tr><td>boardindex</td><td>SS7 signaling board index defined by the board configuration.
Default and only value: None.
Required: Yes</td></tr>
</table>
</td></tr>
</table>

Changing the default H.100 clocking configuration

To change the default H.100 clocking configuration, follow these steps:

Step	Action
1	Use the Vision Console to stop the H.100 Clock Manager service, as described in Working with Programmable Media Platform services.
2	Modify the *priorities.xml* file in the *vx/clockmgr* directory as appropriate. The elements and attributes for this file are described in H.100 clock manager configuration file.
3	Use the Vision Console to restart the H.100 Clock Manager service.

11. Glossary

A

ADTCP: An audio driver that provides a TCP interface to MIOSIP for rendering SSML fragments.

AMR: Adaptive multi-rate; an audio data compression scheme optimized for speech coding. This scheme was adopted by 3GPP and is used in video services.

ASR: Automatic speech recognition; ASR resources, called ASR engines in the MRCP framework, typically enable users of information systems to speak entries rather than punching numbers on a keypad. See also MRCP.

Authorization and Usage Indication interface: XML-over-HTTP mechanism that authorizes call sessions and gathers information for call detail reports.

B

blind transfer: A call transfer in which the originating caller is not announced and is connected directly to destination. In a blind transfer the Vision Server redirects the caller to the callee without remaining in the connection and does not monitor the outcome.

bridge transfer: A blind transfer in which the Vision Server redirects the caller to the callee and remains as a listener.

C

Call Server: Component of the Vision Server that manages call control and routing capabilities.

CCXML: Call Control Extensible Markup Language; a W3C Working Draft standard language for providing telephony call control support for dialog systems, gateways, and conferencing services.

CCXML application definition file: A file that maps individual CCXML applications to number ranges that trigger the execution of those applications.

clock: A periodic reference signal used for synchronization on a transmission facility, such as a telephony bus. See also clock master, clock slave, clock fallback.

clock master: A board that drives the clock signal for a system of boards connected by a bus cable. See also clock slave.

clock slave: A board that derives its clock signal from a bus cable; the clock signal is driven by the bus clock master. See also clock master.

consultation transfer: A call transfer in which the Vision Server initiates a transfer between two parties, but does not stay attached to the call once it is successfully established. The caller remains connected to the Vision Server if the transfer fails.

D

DTMF: Dual tone multi frequency; an inband signaling system that uses two simultaneous voiceband tones for dialing. Also called touchtone. Some times DMTF is used to generally describe any telephony keypad press, even if tones are not generated.

G

G.711: An ITU PCM encoder/decoder specification for mu-law and A-law encoding.

H

H.100 bus: A TDM telephony bus standard for integrating hardware from various PC board vendors. The H.100 specification defines a ribbon cable bus that transports telephony voice data and signaling data across PCI boards. The H.100 bus is an interoperable superset of the H-MVIP and MVIP-90 telephony buses.

H.223: A protocol used to multiplex control and audio and video media on and off of a single DS0 within a trunk.

H.263: An ITU video compression standard. H.263 supports CIF, QCIF, SQCIF, 4CIF and 16CIF resolutions.

H.264: An ITU and ISO video compression standard that compresses video into lower bandwidth compared to H.263 and MPEG-4. H.264 is also called MPEG-4 Part 10.

I

INAP: Intelligent Network Application Part; an SS7 protocol that facilitates building platform-independent, transport-independent, and vendor-independent applications. Such applications include service switching points (SSPs), internet protocol (IP) applications, service control points (SCPs), enhanced services platforms, service circuit nodes, and other custom applications.

ISDN: Integrated services digital network; a standard for providing voice and data telephone service with all digital transmission and message-based signaling.

ISUP: ISDN user part; the SS7 protocol layer that allows for the establishment, supervision, and clearing of circuit-switched connections between two SS7 signaling points, such as central office switches. Despite its name, the ISUP layer is not unique to interconnecting. It is used to manage all types of circuit-switched connections.

ITU: International Telecommunications Union; an international standards body for telecommunications.

IVR: Interactive voice response; a telephony application in which callers interact with programs using recorded or synthesized voice prompts, DTMF digits, or speech recognition to query or deliver information.

M

Media Resource Function: Component of the Programmable Media Platform that provides media processing including record, playback, and interfaces to speech recognition resources. The Media Resource Function is implemented by MIOSIP.

MIB: Management information base; an SNMP collection of objects that represent a managed node. Physically, a list of variables. Logically, a table with rows of variables.

MIOSIP: Implements the Media Resource Function of the Programmable Media Platform. MIOSIP provides SIP call control, media processing over RTP, DTMF generation and recognition, and an MRCP client to automatic speech recognition (ASR) resources.

MPEG-4: An ISO/IEC standard for compressing multimedia data (video, audio, and speech).

MRCP: Media Resource Control Protocol; an application protocol for implementing automatic speech recognition (ASR) and text-to-speech services (TTS). MRCP provides a distributed system of ASR and TTS engines connected over an IP network.

MTP: Message transfer part; the SS7 protocol layers responsible for the reliable, in-sequence delivery of packets between two SS7 signaling points. The MTP functions include message routing, signaling link management, signaling route management, and congestion control.

MVIP-95: Device driver specification for H-MVIP, H.100, and H.110 telephony buses.

N

NETANN: Basic Network Media Services with SIP; an interface that enables applications in a SIP network to locate and invoke basic services on a media server. These services include network announcements, user interaction, and conferencing services. Also called RFC 4240.

O

OSP: Open Settlement Protocol; a European Telecommunications Standards Institute (ESTI) protocol used to exchange authorization, accounting, and usage information for IP telephony.

P

PSTN: Public switched telephone network; a public telephone network.

R

route: A connection path. On the PSTN network, a route is a logical collection of trunks. On the IP network, a route is a destination URL.

RTP: Real time transport protocol; a layer added to the internet protocol (IP) that addressed problems caused when real-time interactive exchanges (such as audio data) are conducted over lines designed to carry packet-switched (connectionless) data.

S

SCCP: Signaling connection control part; an SS7 protocol that provides both connection-oriented and connectionless data transfer over an SS7 network. It extends the service provided by the SS7 MTP layers by adding extended addressing capabilities and multiple classes of service. The SCCP addressing capabilities allow a message to

be addressed to an individual application or database within a signaling point. See also SS7.

SDP: Session description protocol, a protocol that defines a text-based format for describing streaming media sessions and multicast transmissions.

Signaling Server: An optional component of the Vision Server that provides redundant and scalable ISUP signaling.

SIP: Session initiation protocol. An IP signaling and telephony control protocol used mainly for voice over IP calls and multimedia communications. SIP relies on the session description protocol (SDP) for session description and the Real Time Transport Protocol (RTP) for actual transport.

SRGS: Speech Recognition Grammar Specification (SRGS); a syntax for representing the grammars used in speech recognition.

SS7: Signaling system 7; an out-of-band signaling system that provides fast call setup using circuit-switched connections and transaction capabilities for remote database interactions.

SSML: Speech Synthesis Markup Language; a proposed standard for enabling access to the internet using speech. SSML provides a standard way to control various aspects of speech (such as pronunciation, volume, pitch, and rate) over a variety of platforms.

SSML Processor: Component of the Programmable Media Platform that processes SSML requests for audio and text-to-speech.

T

T.38 fax: A standard for real-time fax over IP that makes it possible for fax machines from different vendors to talk to each other over IP networks. The T.38 standard defines how to conduct group 3 facsimile transmission between terminals in which a portion of the transmission path between terminals includes (besides the PSTN or ISDN) an IP network such as the internet.

TCAP: Transaction capabilities application part; an SS7 protocol that provides applications with transaction support over the SS7 network. It enables the exchange of non-circuit related data, such as database queries and responses and remote feature invocation requests between SS7 signaling points. The TCAP layer relies on both the MTP and SCCP layers for message addressing and delivery.

TDM: Time division multiplexing; a technique for transmitting a number of separate data, voice, or video signals simultaneously over one communications medium by quickly interleaving a piece of each signal one after another.

telecom configuration file: File that provides information about the resources that interface with the Call Server and about other elements, such as the number of routes and the circuit selection.

trunk: The physical interface between the telephone network and the Vision Server. In telephone networks, a trunk is a shared connection between two switches. It differs from a line in that it is not dedicated to one subscriber or extension. T1 and E1 trunks carry 24 and 31 circuits, respectively.

TTS: Text-to-speech; a system that converts written language to speech.

V

Vision Console: Web-based configuration tool that configures the Vision Server.

VoiceXML: Voice Extensible Markup Language; a language that enables users to interact with the internet through voice recognition technology.

VoiceXML application configuration file: A file that maps individual VoiceXML applications to number ranges that trigger the execution of those applications.

VoiceXML Interpreter: Component of the Programmable Media Platform that interprets VoiceXML dialogs.

VoiceXML Subsystem: Component of the Programmable Media Platform that provides media processing for VoiceXML applications. The VoiceXML Subsystem consists of the VoiceXML Interpreter, SSML Processor, and Media Resource Function.

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